IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

APPELLANTS:	Chalupper et al	CONFIRMATION NO.: 6889
SERIAL NO.:	10/788,521	GROUP ART UNIT: 2614
FILED:	February 27, 2004	EXAMINER: George C. Monikang
TITLE:	DEVICE AND METHOD TO ADJUST A HEARING DEVICE	

MAIL STOP APPEAL BRIEF-PATENTS
Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313-1450

APPELLANTS' APPEAL BRIEF

S I R:

In accordance with the provisions of 37 C.F.R. §41.37, Appellants herewith submit their main brief in support of the appeal of the above-referenced application.

TABLE OF CONTENTS

TABLE OF AUTHORITIES

REAL PARTY IN INTEREST:

The real party in interest is Siemens Audiologische Technik GmbH, a German company.

RELATED APPEALS AND INTERFERENCES:

There are no related appeals and no related interferences.

STATUS OF CLAIMS:

Claims 21-30 are the subject of the present appeal, and constitute all pending claims of the application. Claims 1-20 were cancelled during previous prosecution before the Examiner. No other claims were present in the application.

STATUS OF AMENDMENTS:

No amendment was filed following the final rejection.

SUMMARY OF CLAIMED SUBJECT MATTER:

Independent claims 20 and 26 are the only independent claims of the application, and are set forth below with exemplary citations to the specification and drawings for all relevant claim limitations therein.

21. A method to automatically adjust a new hearing aid, comprising the steps of:

temporarily bringing a first hearing aid (hearing aid 6 in Figs. 2, 3, 4, 5), having an acoustic input and an acoustic output and that has been worn by a hearing-impaired person, into active communication with a measurement device (plug 7 in the embodiment of Fig. 2; speaker 13 and microphone 14 and measurement device 12 in the embodiments of Figs. 3, 4, 5) that is a separate device from, and is external to, said first hearing aid (p. 8, l. 16-19 and p. 9, l. 4-12);

from a processor (PC 9 in Figs. 2, 3, 4, 5), operating said measurement device to obtain, by said active communication with said first hearing aid, a detected operational characteristic of said first hearing aid that represents overall operation of said first hearing aid between said acoustic input and said acoustic output of said first hearing aid (p. 8, l. 16-26 and p. 9, l. 4-19);

supplying said operational characteristic of said first hearing aid from said measurement device to said processor and, in said processor, automatically analyzing said operational characteristic of said first hearing aid to obtain an analysis result and automatically determining, fro said analysis result, setting parameters for electronic circuitry in a second hearing aid that is to replace said first hearing aid as a new hearing aid to be worn by said hearing-impaired person (p. 8, l. 22-26 and p. 9, l. 13-19);

temporarily placing said second hearing aid (hearing aid 11 in Figs. 2, 3, 4, 5) in active communication with a setting device (plug 10 and modem 8 in the embodiments of Figs. 2, 3; plug 10, modem 8, speaker 15, microphone 16 and measurement device 12 in the embodiment of Fig. 4; speaker and microphone and measurement device 12 in the embodiment of Fig. 5) that is connected to said processor and that is a separate device from, and is external to, said second hearing aid (p. 8, l. 19-21 and p. 9, l. 22-25); and

from said processor, setting said electronic circuitry in said second hearing aid with said setting parameters via said active communication between said setting device and said second hearing aid (p. 8, l. 22-26 and p. 9, l. 23-27).

26. An adjustment device to automatically adjust a new hearing aid, comprising:

a measurement device (plug 7 in the embodiment of Fig. 2; speaker 13 and microphone 14 and measurement device 12 in the embodiments of Figs. 3, 4, 5) that is separate from but configured to temporarily interact with a first hearing aid (hearing aid 6 in Figs. 2, 3, 4, 5), having an acoustic input and an acoustic output and that has been worn by a hearing-impaired person, by active communication between the first hearing aid and the measurement device (p. 8, l. 16-19 and p. 9, l. 4-12);

a processor (PC 9 in Figs. 2, 3, 4, 5) connected to said measurement device that operates said measurement device to obtain, by said active communication with said first hearing aid, a detected operational characteristic of said first hearing aid that represents overall operation of said first hearing aid between said acoustic input and said acoustic output of said first hearing aid (p. 8, l. 16-26 and p. 9, l. 4-19);

said processor being configured to automatically analyze said operational characteristic of said first hearing aid to obtain an analysis result and to determine, from said analysis result, setting parameters for electronic

circuitry in a second hearing aid that is to replace said first hearing aid as a new hearing aid to be worn by said hearing-impaired person (p. 8, l. 22-26 and p. 9, l. 13-19);

a setting device (plug 10 and modem 8 in the embodiments of Figs. 2, 3; plug 10, modem 8, speaker 15, microphone 16 and measurement device 12 in the embodiment of Fig. 4; speaker and microphone and measurement device 12 in the embodiment of Fig. 5) connected to said processor, that is separate from but configured to interact (p. 8, l. 19-21 and p. 9, l. 22-25) with said second hearing aid (hearing aid 11 in Figs. 2, 3, 4, 5) by temporary active communication between said setting device and said second hearing aid; and

said processor operating said setting device to set said electronic circuitry in said second hearing aid with said setting parameters via said active communication between said setting device and said second hearing aid (p. 8, l. 22-26 and p. 9, l. 23-27).

Figs. 1-5 as originally filed are submitted herewith as Exhibit A.

GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL:

The following issues are presented for review in the present appeal:

Whether the subject matter of claims 21, 22, 24-27 and 29-30 would have been obvious to a person of ordinary skill in the field of designing systems for setting parameters in hearing aids, under the provisions of 35 U.S.C. §103(a), based on the teachings of United States Patent No. 6,035,050 (Weinfurtner et al., Exhibit B) in

view of the teachings of United States Patent Application Publication No. 2002/0078161 (Cheng, Exhibit C); and

Whether the subject matter of claims 23 and 28 would have been obvious to a person of ordinary skill in the field of designing systems for setting parameters in hearing aids, under the provisions of 35 U.S.C. §103(a), based on the teachings of Weinfurtner et al. and Cheng and further in view of the teachings of United States Patent No. 5,825,894 (Shennib, Exhibit D).

ARGUMENT:

Rejection of Claims 21, 22, 24-27 and 29-30 under 35 U.S.C. §103(a) based on Weinfurtner et al and Cheng

The method of independent claim 21 and the adjustment device of independent claim 26 are for the purpose of automatically setting a second hearing aid, which is to replace a first hearing aid worn by a user, with the same operating parameters that are in existence in the first hearing aid. This is for the purpose of alleviating the time-consuming procedure, often requiring the involvement of a trained audiologist, in order to set a replacement hearing aid for a user in a manner that conforms to the current hearing aid being worn by the user.

As explained in the present specification, after a user has worn a particular hearing aid for a period of time, the user becomes accustomed to and comfortable with the settings for that hearing aid. If and when it is necessary to replace the person's current hearing aid with a replacement hearing aid, it is desirable, at least initially, to set the replacement hearing aid parameters to the same parameter settings as the user became accustomed to by wearing and using the first hearing aid.

Conventionally, this was done by an audiologist reading out the parameter settings from the first hearing and transferring the same setting to a second hearing aid, through a transfer procedure requiring the manual involvement of the audiologist, and individual parameter-by-parameter setting of the replacement hearing aid.

This conventional procedure is not needed in accordance with the subject matter of the claims on appeal, by virtue of the first hearing aid being temporarily brought into communication with a device that reads out and stores the parameter settings, and then temporarily bringing the second hearing device into communication with the setting device, so that those same parameters are then automatically used to set the replacement (second) hearing device.

Appellants submit that the Weinfurtner et al. reference discloses setting parameters in only a single hearing aid, which is temporarily fitted with an auxiliary module 20 so as to place the hearing aid into communication with a control module. Via the control module and the temporarily-fitted auxiliary module, the settings of the hearing aid can be remotely changed.

There is no disclosure in the Weinfurtner et al. reference to make use of such a procedure for any purpose other than setting the currently-worn hearing aid.

The Cheng reference cited by the Examiner has nothing whatsoever to do with hearing aids, and is concerned with allowing non-UPnP (Universal Plug and Play) devices compatible with a conventional UPnP controller. As is clear from claim 1 of the Cheng reference, the so-called bridge device or enabling device is merely a device that effects a transformation between a UPnP-compatible protocol and a non-UPnP-compatible protocol. There is no "setting" of any other device involved, the

bridging device merely makes one device usable with another, without internally changing either device.

The only reason why a system of the type disclosed in the Cheng reference might be used in combination with a hearing aid system would be if two hearing aids, or a hearing aid and a hearing aid controller, were, for some reason, not compatible with each other. Even if the Cheng reference were used for that purpose in such a context, there still is no teaching in either of the Cheng reference or the Weinfurtner et al. reference to make use of the transformation procedure disclosed in the Cheng reference for the purpose of automatically setting parameters in a replacement hearing aid to the same settings as were used in a currently-worn hearing aid, as disclosed and claimed in the present application.

Appellants respectfully submit that it is clearly the case that the Examiner has relied on the Cheng reference, and, more likely, that the Examiner has even located the Cheng reference, only after reading Appellants' disclosure and being guided by the contents of Appellants' disclosure. This is clearly an impermissible manner to formulate a rejection under 35 U.S.C. §103(a).

More importantly, however, for the reasons noted above, even if the Weinfurtner et al. system were modified in accordance with the teachings of Cheng, the subject matter of the claims of the present application still would not result.

Conversely, if a person of ordinary skill in the field of hearing aid design, seeking to solve the aforementioned problem of quickly and easily setting a replacement hearing aid to the same parameters as a currently-worn hearing aid, had the insight to make use of techniques relating to transforming a non- UPnP

device for use with a UPnP device, this would be an insight supporting patentability, rather than a reason for precluding patentability.

In response to these arguments that were made during prosecution, the Examiner stated, in the Final Rejection dated April 19, 2010, that the Weinfurtner reference discloses, in Fig. 2, that there "could be another hearing aid . . . where the parameters could be set by control module 40." Appellants respectfully submit that the Weinfurtner et al. reference does not provide any such disclosure.

Figure 2 of the Weinfurtner et al. reference merely shows a different embodiment from the embodiment of Fig. 1. In the embodiment of Fig. 1 of Weinfurtner et al., a behind-the-ear hearing aid 10 is shown as being fitted to the auxiliary module 20, and in the embodiment of Fig. 2 an in-the-ear hearing aid 10' is shown as being electrically connected to the auxiliary module 20. In each embodiment, however, it is only the hearing aid 10, or the hearing aid 10', that is set with parameters remotely via the control module 40. There is no disclosure whatsoever in Weinfurtner et al. of transferring parameters between these two different types of hearing aids, and in fact those of ordinary skill in the field of hearing aid design would know that this is not a realistic possibility because the microphone, that detects incoming audio signals, and the earphone, that emits an output audio signal to the hearing aid wearer, are each located at different locations in the case of a behind-the-ear hearing aid as opposed to an in-the-ear hearing aid. Therefore, those of ordinary skill in the field of hearing aid design would know in advance that parameter settings for a behind-the-ear hearing aid could not be expected to be automatically transferred for use in an in-the-ear hearing aid. Even if basic settings might be expected to be similar for these two different types of hearing aids with

regard to the same hearing-impaired person, those of ordinary skill in the field of hearing aid design know that those basic settings would still require adjustment, in view of the different locations of the microphone and the earphone, respectively, in those two different types of hearing aids.

Also in response to the aforementioned arguments that were made during previous prosecution, the Examiner, in the April 9, 2010 Final Rejection stated that the UPnP message enabling device disclosed in the Cheng reference is capable of "preprocessing in one device and sending that preprocess to another device where it can be executed." The Examiner stated that, for example, a noise compensation algorithm in one computer could be sent to another computer where the noise compensation algorithm can be utilized and executed the same way it was in the previous computer. The Examiner stated that based on that analysis, the combination of Weinfurtner et al. and Cheng is not hindsight.

Appellants respectfully submit this statement of the Examiner is evidence in support of the Appellants' non-obviousness position, rather than the Examiner's position regarding obviousness. Appellants' acknowledge that one basic premise of a "plug and play" system is that programs in one component of the system can be transferred in tact to another component of the system, and then used in that other component. This transfer is based on the further premise, however, that no adjustment or modification of the transferred program will be necessary for use in the device to which the program is transferred. Therefore, the message enabling device disclosed in the Cheng reference merely allows slightly differing component formats to be made compatible with each other, so that the transfer can take place. The actual program that is transferred, however, is transferred intact from one component

of the system to the other. This is in contrast to the explicit language of each of independent claims 21 and 26, which requires that the processor automatically analyze an operational characteristic of the first hearing aid, in order to obtain an analysis result, and then makes use of that analysis result to automatically determine setting parameters for electronic circuitry in the second hearing aid. Therefore, the subject matter disclosed and claimed in the present application clearly does not simply transfer one set of parameters intact from the first hearing aid to the second hearing aid, but instead performs an analysis on the parameters measured in the first hearing aid, so as to determine appropriate setting parameters therefrom for use in the second hearing aid.

The fact that the Examiner has ignored this explicit language in claims 21 and 26 substantiates Appellants' conclusion that the Examiner has selected and combined the Weinfurtner et al. and Cheng et al. references solely after reading Appellants' disclosure, and has not undertaken a careful comparison of the actual claim language with regard to the teachings of those respective references.

The Federal Circuit stated in *In re Lee* 227 F.3d 1338, 61 U.S.P.Q. 2d 1430 (Fed. Cir. 2002):

> "The factual inquiry whether to combine references must be thorough and searching. ...It must be based on objective evidence of record. This precedent has been reinforced in myriad decisions, and cannot be dispensed with."

Similarly, quoting *C.R. Bard, Inc. v. M3 Systems, Inc.*, 157 F.3d 1340, 1352, 48 U.S.P.Q. 2d 1225, 1232 (Fed. Cir. 1998), the Federal Circuit in *Brown & Williamson Tobacco Court v. Philip Morris, Inc.*, 229 F.3d 1120, 1124-1125, 56 U.S.P.Q. 2d 1456, 1459 (Fed. Cir. 2000) stated:

> [A] showing of a suggestion, teaching or motivation to combine the prior art references is an 'essential component of an obviousness holding'.

In *In re Dembiczak*, 175 F.3d 994,999, 50 U.S.P.Q. 2d 1614, 1617 (Fed. Cir. 1999) the Federal Circuit stated:

> Our case law makes clear that the best defense against the subtle but powerful attraction of a hindsight-based obviousness analysis is rigorous application of the requirement for a showing of the teaching or motivation to combine prior art references.

Consistently, in *In re Rouffet*, 149 F.3d 1350, 1359, 47 U.S.P.Q. 2d 1453, 1459 (Fed. Cir. 1998), the Federal Circuit stated:

> [E]ven when the level of skill in the art is high, the Board must identify specifically the principle, known to one of ordinary skill in the art, that suggests the claimed combination. In other words, the Board must explain the reasons one of ordinary skill in the art would have been motivated to select the references and to combine them to render the claimed invention obvious.

In *Winner International Royalty Corp. v. Wang*, 200 F.3d 1340, 1348-1349, 53 U.S.P.Q. 2d 1580, 1586 (Fed. Cir. 2000), the Federal Circuit stated:

> Although a reference need not expressly teach that the disclosure contained therein should be combined with another, ... the showing of combinability, in whatever form, must nevertheless be clear and particular.

Lastly, in *Crown Operations International, Ltd. v. Solutia, Inc.*, 289 F.3d 1367, 1376, 62 U.S.P.Q. 2d 1917 (Fed. Cir. 2002), the Federal Circuit stated:

> There must be a teaching or suggestion within the prior art, within the nature of the problem to be solved, or within the general knowledge of a person of ordinary skill in the field of the invention, to look to particular sources, to select particular elements, and to combine them as combined by the inventor.

Appellants submit that the decision of the United States Supreme Court in *KSR International Co. v. Teleflex Inc.*, 550 U.S. 398, 127 S.Ct. 1727, 82 U.S.P.Q. 2d 1385 (2007), and the United States Patent and Trademark Office guidelines for

applying that decision, support the position of the Appellants. That decision, although stating that it is not always required to point to a specific teaching in a prior art reference in order to substantiate a rejection under 35 U.S.C. §103(a), by no means approved ignoring the above long-standing precedent, and certainly did not represent a blanket overruling of that precedent. In the *KSR* decision, the Supreme Court stated, *under certain circumstances*, it may not be necessary to point to a specific passage in a prior art reference as evidence of motivation, guidance or inducement in order to modify that reference in a manner that obviates the patent claim in question. The Supreme Court stated that if a person of ordinary skill in the art can implement a *predictable variation* and would see the benefit of doing so, Section 103(a) likely bars patentability.

Nevertheless, the Supreme Court also stated that the requirement to find a teaching, suggestion or motivation in the prior art "captures a helpful insight." The Supreme Court stated that although common sense directs caution as to a patent application claiming as innovation the combination of two known devices according to their established functions, it can be important to identify a reason that would have prompted a person of ordinary skill in the art to combine the elements as the new invention does. The Supreme Court, however, stated that not every application requires such detailed reasoning. The Supreme Court stated that helpful insights need not become rigid and mandatory formulas. The Supreme Court only stated that if the requirement to find a teaching, suggestion or motivation is required in such a rigid, formulaic manner, it is then inconsistent with the precedence of the Supreme Court. In fact, the Supreme Court stated that since the "teaching, suggestion or motivation" test was devised, the Federal Circuit doubtless has applied it in accord

with these principles in many cases. The Supreme Court stated there is no necessary inconsistency between this test and an analysis conducted under the standards of *Graham v. Deere*. The Supreme Court stated the only error is transforming this general principle into a "rigid rule limiting the obviousness inquiry."

Therefore, Appellants submit this decision of the Supreme Court does not in any manner approve, much less require, the absence of a rigorous evidentiary investigation on the part of the Examiner in order to substantiate most rejections under 35 U.S.C. §103(a). Only under the somewhat unusual, and very limited, circumstances outlined by the Supreme Court in the *KSR* decision might the Supreme Court excuse the absence of such a rigorous evidentiary investigation in reaching a conclusion of obviousness under 35 U.S.C. §103(a).

This view of the *KSR* decision has been substantiated by the United States Court of Appeals for the Federal Circuit in *Takeda Chemical Industries Limited v. Alphapharm Pty. Ltd.*, 492 F.3d 1350, 83 U.S.P.Q.2d, 169 (Fed. Cir. 2007), which was one of the earliest decisions of the Federal Circuit after the *KSR* decision was decided by the Supreme Court. The *Takeda* decision concerned a chemical patent that was the subject of an infringement lawsuit, and which was attacked by the infringer on the basis of the claimed subject matter being "obvious to try." After acknowledging that the *KSR* decision held that the teaching-suggestion-motivation test should not be applied rigidly, the Federal Circuit stated that the *KSR* decision actually recognized the value of that test in determining whether the prior art provided a *reason* for one of skill in the art to make the claimed combination. The Federal Circuit stated this is consistent with the Federal Circuit precedent in *In re Dillon*, 919 F.2d 688 (Fed. Cir. 1990) and in *In re Deuel*, 51 F.3d 1552 (Fed. Cir.

1995). The Federal Circuit stated that in cases involving new chemical compounds, it remains necessary to identify some reason that would have led a chemist to modify a known compound in a particular manner to establish *prima facie* obviousness of the new claimed compound. In the *Takeda* decision, the Federal Circuit stated:

> The *KSR* Court recognized that "[w]hen there is a design need or market pressure to solve a problem and there are a finite number of identified, predictable solutions, a person of ordinary skill has good reason to pursue the known options within his or her technical grasp," *KSR*, 127 S.Ct. at 1732. In such circumstances, "the fact that a combination was obvious to try might show that it would obvious under §103." *id.* that is not the case here. Rather than identify predictable solutions for antidiabetic treatment, the prior art disclosed a broad selection of compounds, any one of which could have been selected as a lead compound for further investigation.

For the foregoing reasons, Appellants respectfully submit the Examiner has not properly substantiated the rejection of independent claims 1 and 26 under 35 U.S.C. §103(a) based on Weinfurtner et al. and Cheng.

Appellants' above arguments are applicable as well to the rejection of claims 22, 24, 25, 27, 29 and 30, respectively depending from independent claims 1 and 26, in view of the statements of the Federal Circuit in *In Re Fine*, 837 F.2d 1071, 1076 (Fed. Cir. 1988) (If an independent claim is not obvious under U.S.C. §103(a), then any claim depending therefrom is non-obvious).

Rejection of Claims 23 and 28 under 35 U.S.C. §103(a) based on Weinfurtner et al, Cheng and Shennib

The Examiner relied on the Shennib reference as disclosing a hearing aid that is capable of analyzing input signals and generating an ear canal transfer function for the user. Appellants acknowledge that the Shennib reference provides this general teaching, but in substantiating the rejection of claims 23 and 28 based on Weinfurtner et al., Cheng and Shennib, the Examiner stated that the "combined

teachings" of Weinfurtner et al. and Cheng disclose the use of a measurement device comprising a speaker and microphone, and Appellants respectfully submit that no such teaching is present in the "combined teachings" of Weinfurtner et al. and Cheng.

In the rejection of independent claims 21 and 26, the Examiner relied on the auxiliary module 20 of Weinfurtner et al. as allegedly corresponding to the measurement device in those independent claims. In each of dependent claims 23 and 28, this measurement device is further defined as including a speaker and a microphone. The auxiliary module 20 disclosed in the Weinfurtner et al. reference, however, is clearly a plug-in component, and there is no disclosure whatsoever in Weinfurtner et al. that the auxiliary module includes, or even could include, a speaker and a microphone. In column 4, line 44 of Weinfurtner et al., it is stated that the auxiliary module 20, in the embodiment of Fig. 1, is "detachably plugged" to the hearing aid 10 that is worn behind the ear. At column 4, line 45, it is further stated that the hearing aid 10 and the auxiliary module 20 are "electrically connected with one another via contact surfaces." This clearly does not and cannot involve a speaker/microphone arrangement of the type set forth in claims 21 and 28.

In the embodiment of Weinfurtner et al. shown in Fig. 2, concerning the behind-the-ear hearing aid 10', Weinfurtner et al. state at column 4, lines 55-59, that the auxiliary module is connected to the hearing aid 10' via a connection line 12 that is detachably connected to the hearing aid 10' by means of known connection elements (plugs/sockets, etc.), as are used for the hard-wired programming of hearing aids. This again clearly teaches away from the use of a microphone/speaker arrangement.

Therefore, even if the Weinfurtner et al./Cheng combination were further modified in accordance with the teachings of Shennib, the subject matter of claims 23 and 28 still would not result. Therefore, neither of those claims would have been obvious to a person of ordinary skill in the field of designing systems for setting parameters in hearing aids, under the provisions of 35 U.S.C. §103(a), based on the teachings of Weinfurtner et al., Cheng and Shennib.

CONCLUSION:

For the foregoing reasons, Appellants respectfully submit the Examiner is in error in fact and in law in rejecting the claims on appeal. Reversal of those rejections is proper, and the same is respectfully requested.

This Appeal Brief is accompanied by electronic payment in the amount of $540.00 for the requisite fee.

The Commissioner is hereby authorized to charge any additional fees which may be required, or to credit any overpayment to Account No. 501519.

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CLAIMS APPENDIX

21. A method to automatically adjust a new hearing aid, comprising the steps of:

temporarily bringing a first hearing aid, having an acoustic input and an acoustic output and that has been worn by a hearing-impaired person, into active communication with a measurement device that is a separate device from, and is external to, said first hearing aid;

from a processor, operating said measurement device to obtain, by said active communication with said first hearing aid, a detected operational characteristic of said first hearing aid that represents overall operation of said first hearing aid between said acoustic input and said acoustic output of said first hearing aid;

supplying said operational characteristic of said first hearing aid from said measurement device to said processor and, in said processor, automatically analyzing said operational characteristic of said first hearing aid to obtain an analysis result and automatically determining, fro said analysis result, setting parameters for electronic circuitry in a second hearing aid that is to replace said first hearing aid as a new hearing aid to be worn by said hearing-impaired person;

temporarily placing said second hearing aid in active communication with a setting device that is connected to said processor and that is a separate device from, and is external to, said second hearing aid; and

from said processor, setting said electronic circuitry in said second hearing aid with said setting parameters via said active communication between said setting device and said second hearing aid.

22. A method as claimed in claim 21 wherein said first hearing aid has a memory in which setting parameters for electronic circuitry in said first hearing aid are stored, and wherein the step of obtaining said operational characteristic from said first hearing aid comprises reading out said setting parameters from said memory of said first hearing aid and supplying said setting parameters read from the memory of the first hearing aid to said processor, and wherein said second hearing aid has a memory connected to said electronic circuitry of said second hearing aid, and wherein the step of setting said electronic circuitry in said second hearing aid with said setting parameters determined from said operational characteristic of said first hearing aid comprises entering the setting parameters read from said memory of said first hearing aid into said memory of said second hearing aid.

23. A method as claimed in claim 21 wherein said measurement device comprises a speaker and a microphone, and wherein the step of obtaining said operational characteristic of said first hearing aid comprises emitting an acoustic signal from said speaker into said acoustic input of said first hearing aid and detecting an acoustic signal with said microphone from said acoustic output of said first hearing aid, and wherein the step of automatically analyzing said operational characteristic of said first hearing aid comprises automatically identifying, as said analysis result, a transfer function of said first hearing aid, between said acoustic input and said acoustic output, as a ratio of said signal supplied to said acoustic

input of said first hearing aid and said signal emitted from said acoustic output of said first hearing aid.

24. A method as claimed in claim 21 wherein said measurement device is a first measurement device, and comprising placing said second hearing aid in active communication with a second measurement device, and operating said second measurement device from said processor to obtain an operational characteristic representing overall operation of said second hearing aid between an acoustic input thereof and an acoustic output thereof.

25. A method as claimed in claim 24 comprising, in said processor, automatically analyzing said operational characteristic of said second hearing aid and, from said operational characteristic of said second hearing aid, automatically determining modified setting parameters and, from said processor, re-adjusting said second hearing aid according to said modified setting parameters via said active communication between said second hearing aid and said setting device.

26. An adjustment device to automatically adjust a new hearing aid, comprising:

a measurement device that is separate from but configured to temporarily interact with a first hearing aid, having an acoustic input and an acoustic output and that has been worn by a hearing-impaired person, by active communication between the first hearing aid and the measurement device;

a processor connected to said measurement device that operates said measurement device to obtain, by said active communication with said

first hearing aid, a detected operational characteristic of said first hearing aid that represents overall operation of said first hearing aid between said acoustic input and said acoustic output of said first hearing aid;

said processor being configured to automatically analyze said operational characteristic of said first hearing aid to obtain an analysis result and to determine, from said analysis result, setting parameters for electronic circuitry in a second hearing aid that is to replace said first hearing aid as a new hearing aid to be worn by said hearing-impaired person;

a setting device connected to said processor, that is separate from but configured to interact with said second hearing aid by temporary active communication between said setting device and said second hearing aid; and

said processor operating said setting device to set said electronic circuitry in said second hearing aid with said setting parameters via said active communication between said setting device and said second hearing aid.

27. An adjustment device as claimed in claim 26 wherein said first hearing aid has a memory in which setting parameters for electronic circuitry in said first hearing aid are stored, and wherein said processor operates said measurement device to obtain said operational characteristic from said first hearing aid by reading out said setting parameters from said memory of said first hearing aid, and wherein said second hearing aid has a memory connected to said electronic circuitry of said

second hearing aid, and wherein said processor operates said setting device to said electronic circuitry of said second hearing aid with said setting parameters determined from said operational characteristic of said first hearing aid by entering the setting parameters read from said memory of said first hearing aid into said memory of said second hearing aid.

28. An adjustment device as claimed in claim 26 wherein said measurement device comprises a speaker and a microphone, and wherein said processor operates said measurement device to obtain said operational characteristic of said first hearing aid by causing emission of an acoustic signal from said speaker into said acoustic input of said first hearing aid and by detecting an acoustic signal with said microphone from said acoustic output of said first hearing aid, and wherein said processor automatically analyzes said operational characteristic of said first hearing aid by automatically identifying, as said analysis result, a transfer function of said first hearing aid, between said acoustic input and said acoustic output, as a ratio of said signal supplied to said acoustic input of said first hearing aid and said signal emitted from said acoustic output of said first hearing aid.

29. An adjustment device as claimed in claim 26 wherein said measurement device is a first measurement device, and comprising a second measurement device that interacts with said second hearing aid by active communication between said second measurement device and said second measurement device, and operates said second measurement device to obtain an operational characteristic representing overall operation of said second hearing aid between an acoustic input and an acoustic output of said second hearing aid.

30. An adjustment device as claimed in claim 29 wherein said processor is configured to automatically analyze said operational characteristic of said second hearing aid and, from said operational characteristic of said second hearing aid, automatically determine modified setting parameters and to operate said setting device to re-adjusting said second hearing aid according to said modified setting parameters via said active communication between said second hearing aid and said setting device.

EVIDENCE APPENDIX

Exhibit A: Figures 1, 2, 3, 4 and 5 filed with the original Application on February 27, 2004

Exhibit B: United States Patent No. 6,035,050 (Weinfurtner et al.) – cited in the final rejection dated April 9, 2010

Exhibit C: United States Patent Application Publication No. 2002/0078161 (Cheng) – cited in the final rejection dated April 9, 2010

Exhibit D. United States Patent No. 5,825,894 (Shennib) – cited in the final rejection dated April 9, 2010

RELATED PROCEEDINGS APPENDIX

None.

CH2\8951788.1

FIG 1

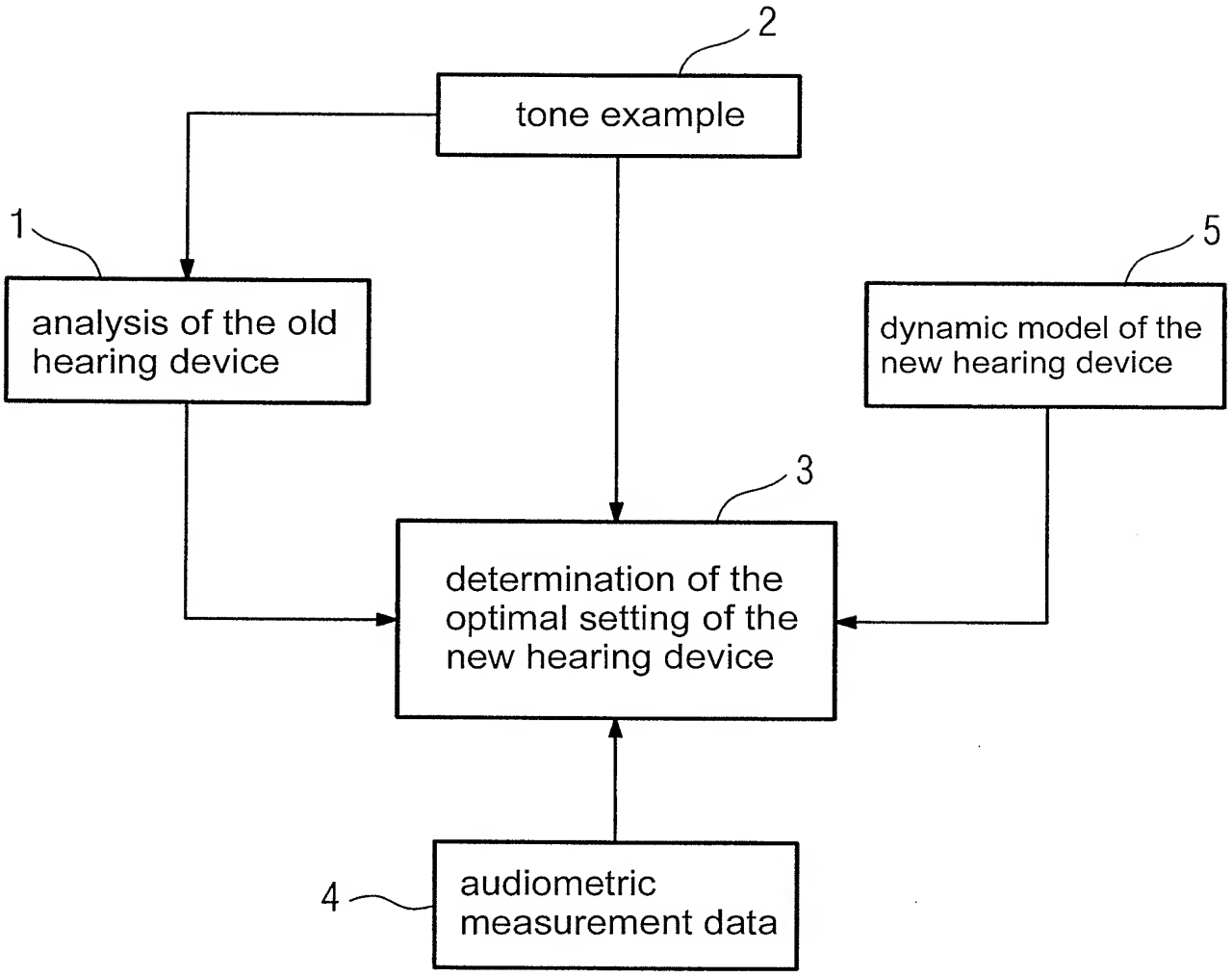


FIG 2

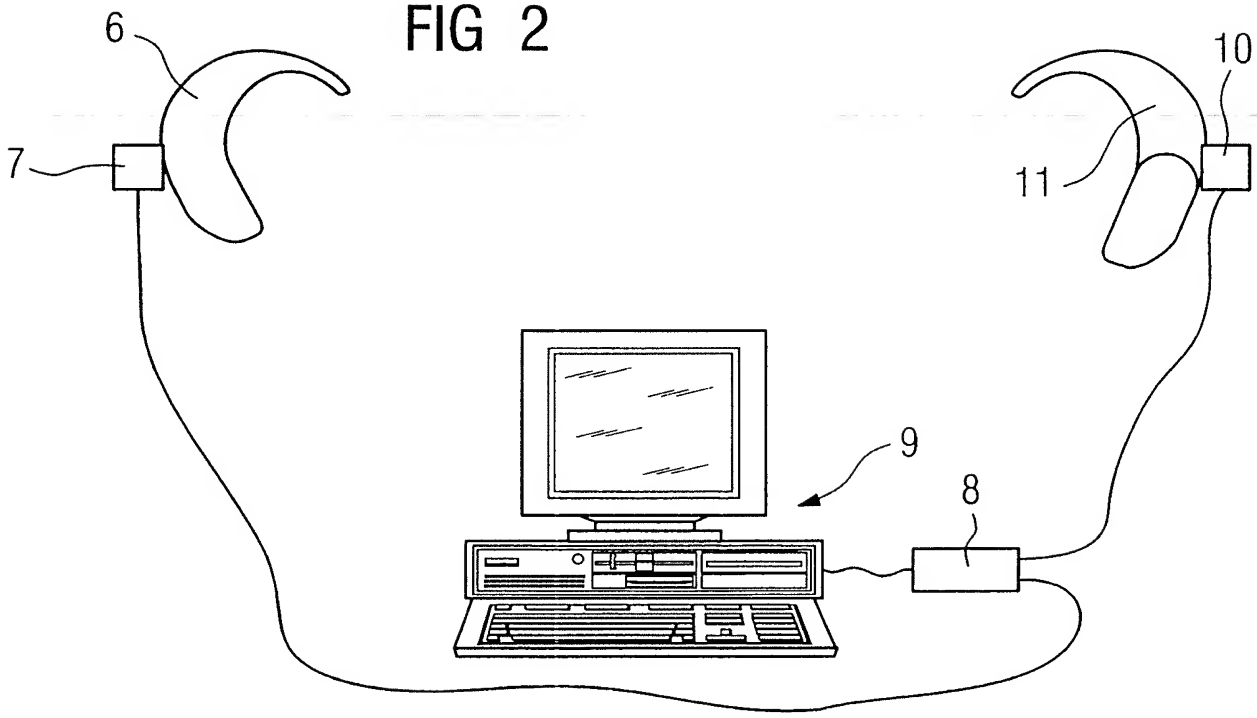


FIG 3

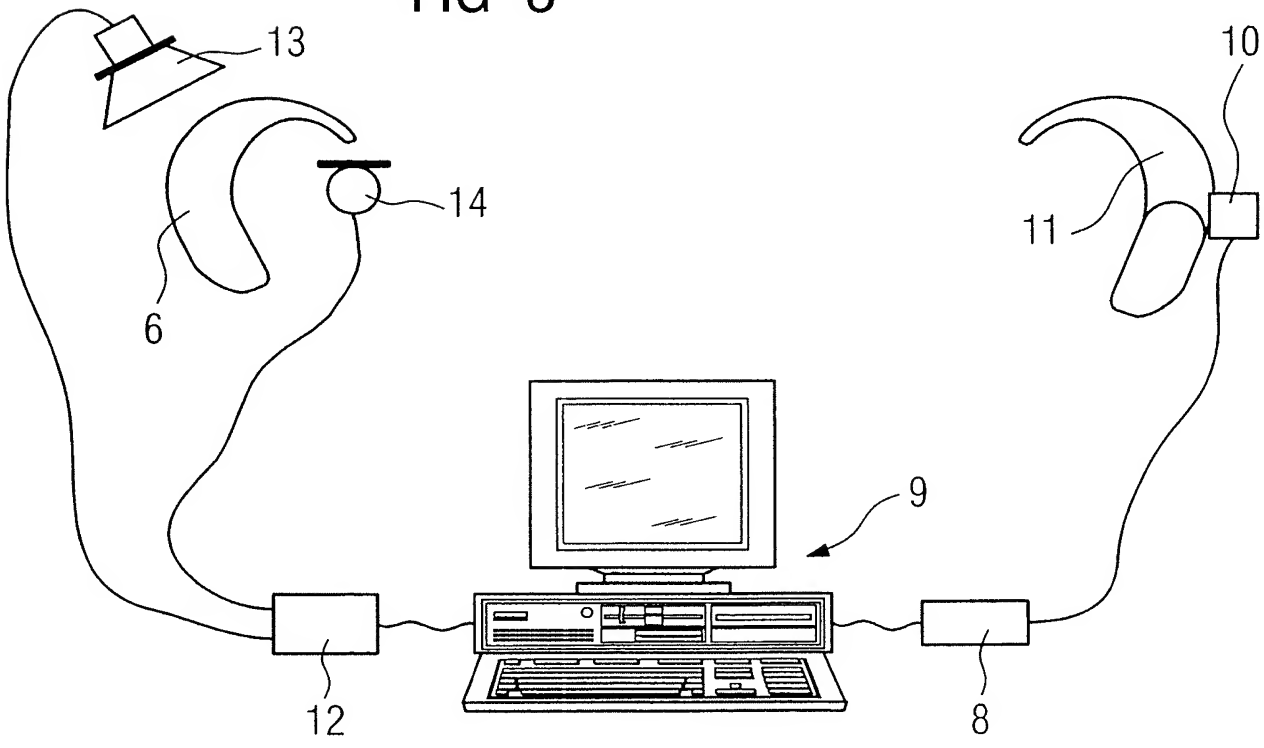


FIG 4

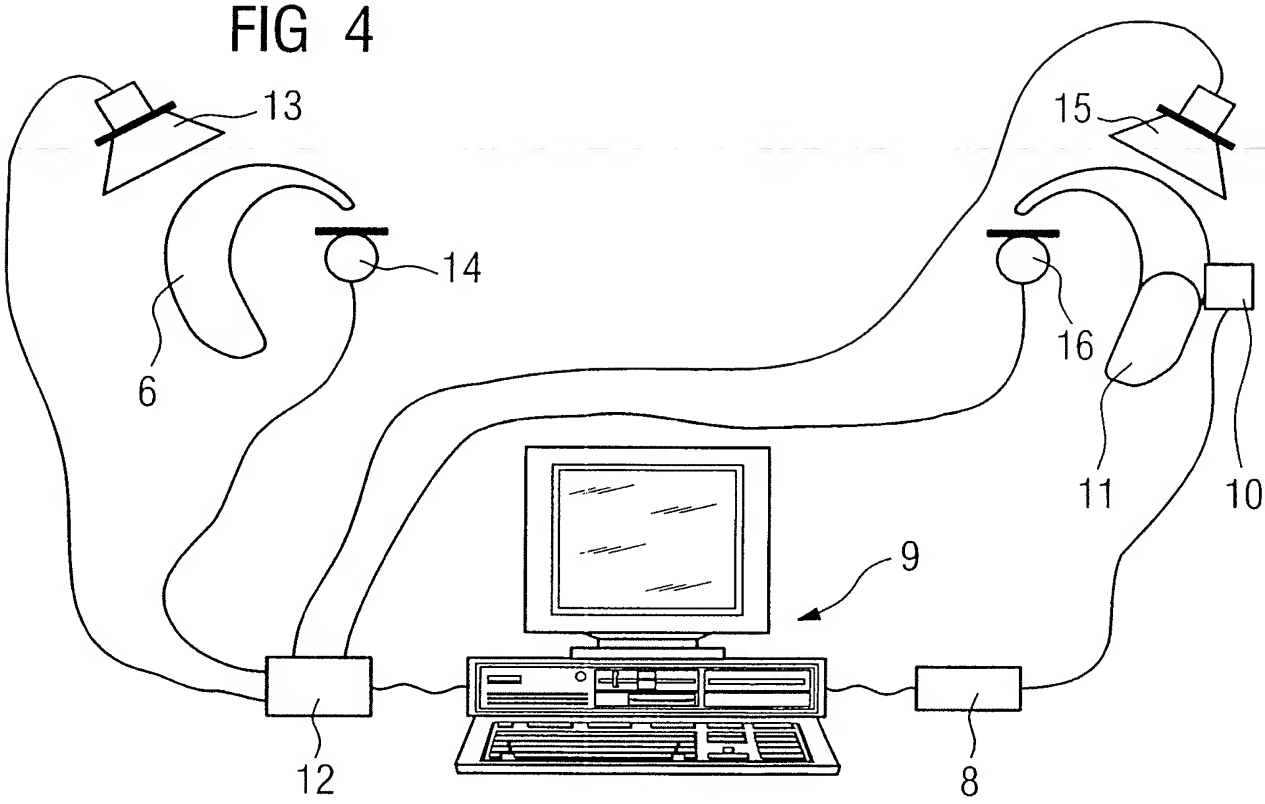
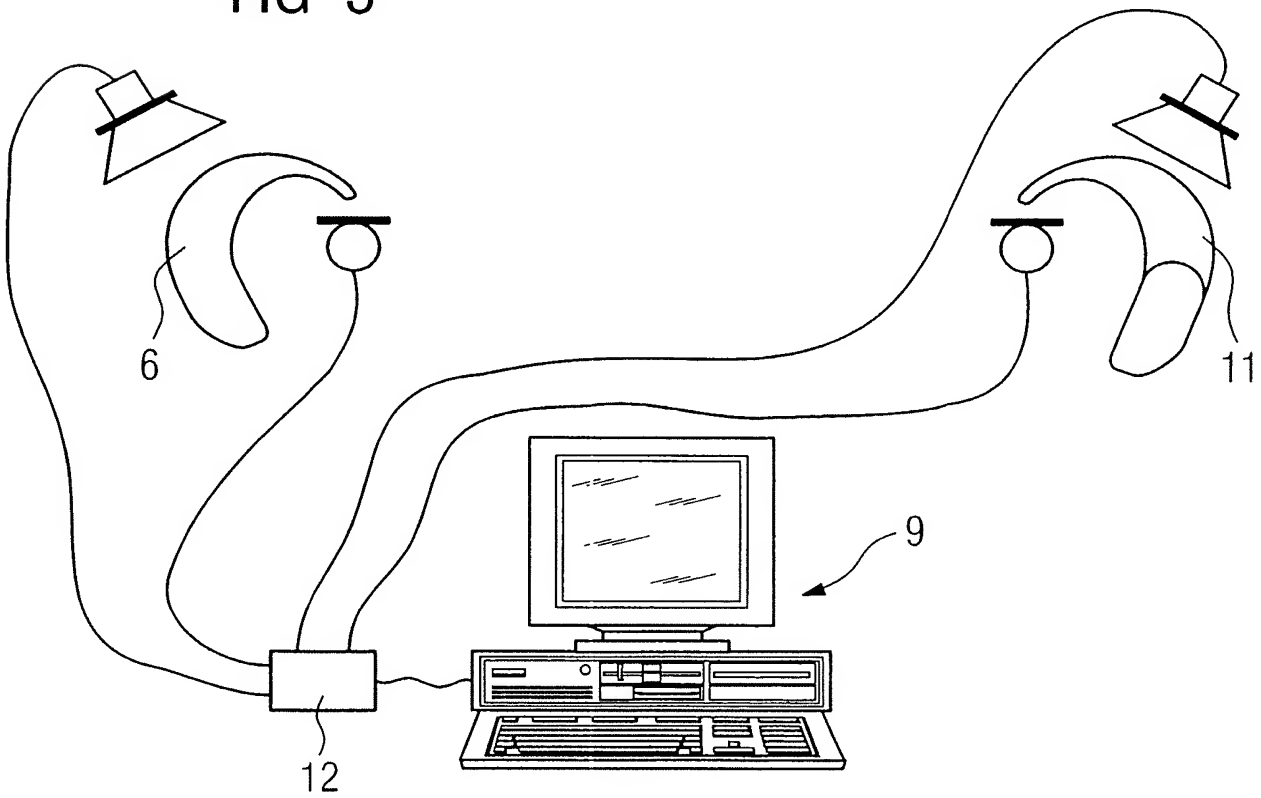


FIG 5



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United States Patent [19]

Weinfurtner et al.

[11] Patent Number: 6,035,050

[45] Date of Patent: Mar. 7, 2000

[54] **PROGRAMMABLE HEARING AID SYSTEM AND METHOD FOR DETERMINING OPTIMUM PARAMETER SETS IN A HEARING AID**

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[73] Assignee: **Siemens Audiologische Technik GmbH**, Erlangen, Germany

[21] Appl. No.: **08/874,456**

[22] Filed: **Jun. 17, 1997**

[30] **Foreign Application Priority Data**

Jun. 21, 1996 [EP] European Pat. Off. 96110067

[51] **Int. Cl.**[7] **H04R 25/00**

[52] **U.S. Cl.** **381/313**; 381/314

[58] **Field of Search** 381/320, 321, 381/312, 314, 323

[56] **References Cited**

U.S. PATENT DOCUMENTS

3,989,904	11/1976	Rohrer et al.	179/107 FD
4,259,547	3/1981	Valley et al. .	
4,425,481	1/1984	Mansgold et al.	179/107 FD
4,947,432	8/1990	T Pholm	381/68.2
4,989,251	1/1991	Mansgold	381/315
5,202,927	4/1993	T pholm .	
5,303,306	4/1994	Brillhart .	
5,404,407	4/1995	Weiss	381/68
5,604,812	2/1997	Meyer	381/68.2
5,610,988	3/1997	Miyahara	381/68.4
5,710,820	1/1998	Martin et al.	381/68.4
5,721,783	2/1998	Anderson	381/68.6

FOREIGN PATENT DOCUMENTS

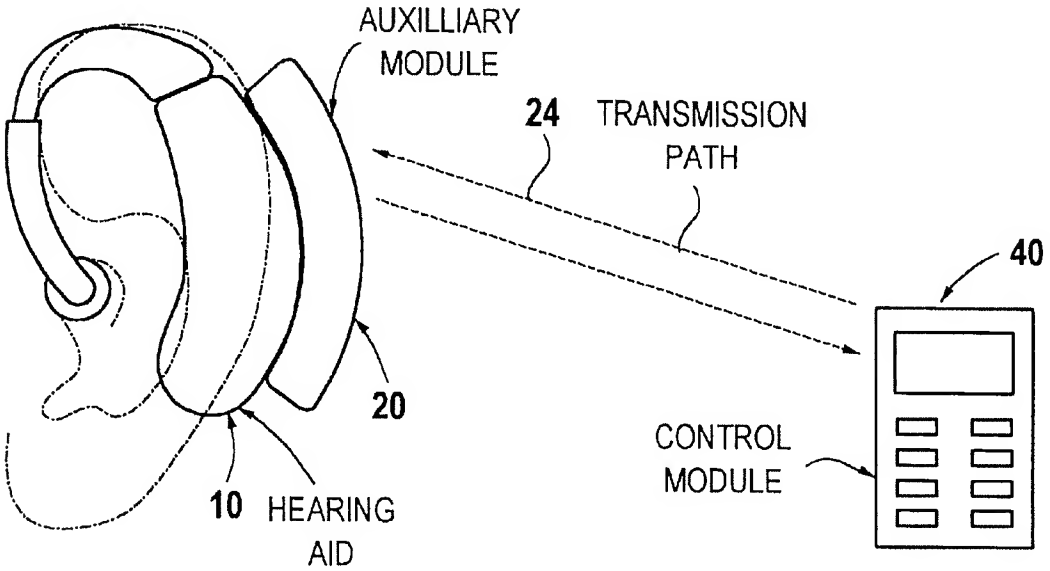
0 712 263	11/1994	European Pat. Off. .
OS 43 08 157	9/1994	Germany .
OS 43 40 817	6/1995	Germany .

Primary Examiner—Curtis A. Kuntz
Assistant Examiner—Dionne N. Harvey
Attorney, Agent, or Firm—Hill & Simpson

[57] **ABSTRACT**

A hearing aid system with a hearing aid has a matching arrangement with a first memory for several parameter sets available for selection for each of several hearing situations, an input unit for selecting a current hearing situation and for selecting one of the several parameter sets available for this hearing situation, and a second memory for allocation data that identify the parameter sets selected for each hearing situation. For the determination of an optimal parameter set for each of several hearing situations, an optimal user-specific parameter set is allocated to each hearing situation as it arises during an optimization phase. After the optimization phase, the allocation data are evaluated for the determining an optimal parameter set for each hearing situation. This parameter set is then permanently programmed as the parameter set which will be called to set the transmission characteristics of the hearing aid whenever the hearing situation allocated thereto occurs.

15 Claims, 3 Drawing Sheets



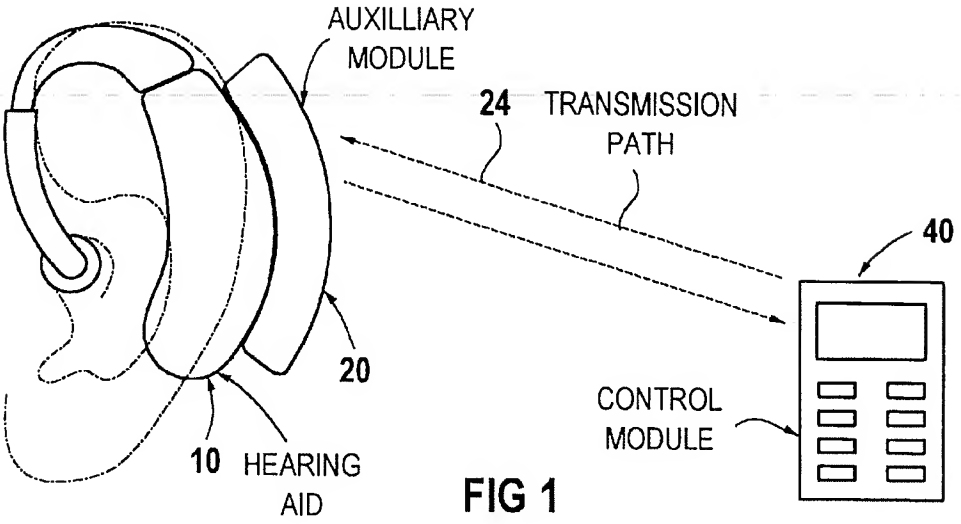


FIG 1

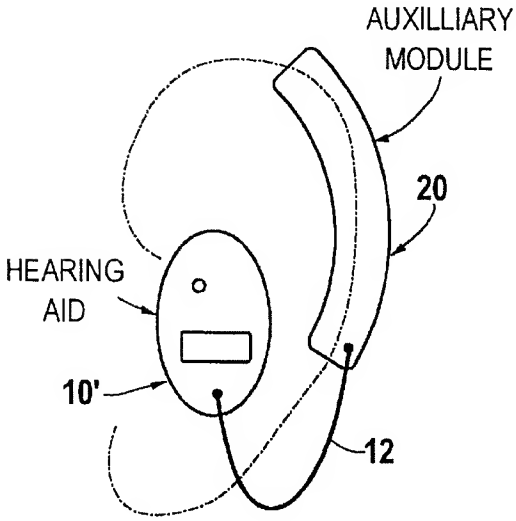


FIG 2

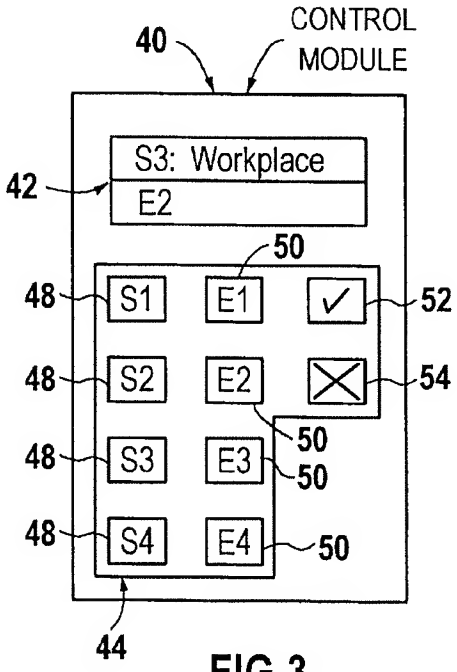


FIG 3

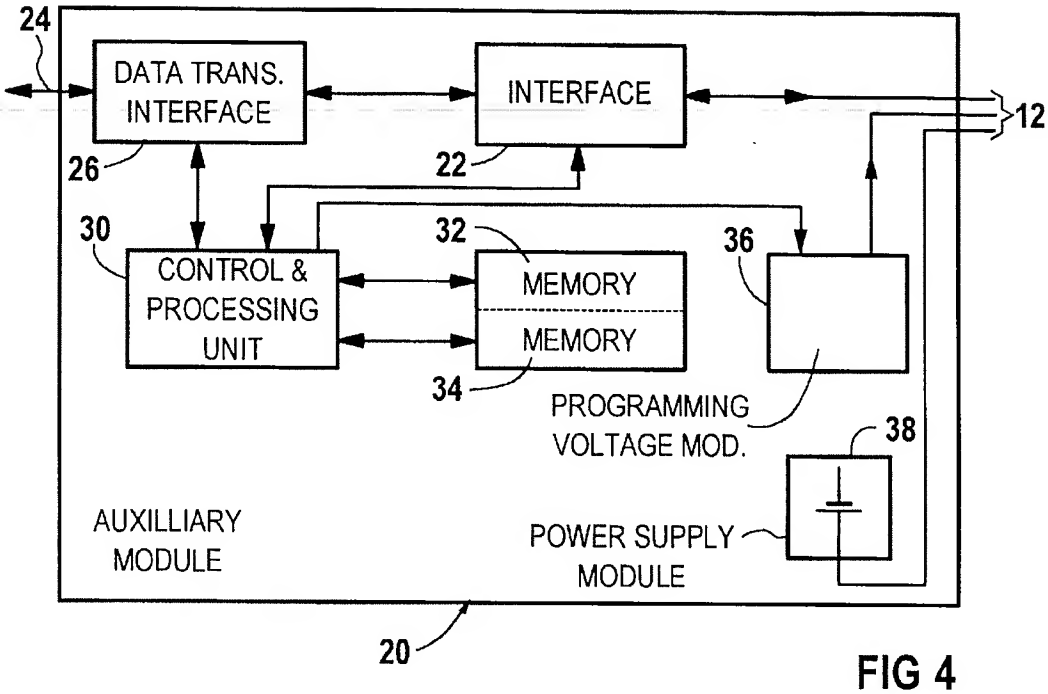

24
DATA TRANS. INTERFACE
INTERFACE
12
26
22
30
CONTROL & PROCESSING UNIT
32
MEMORY
MEMORY
34
36
PROGRAMMING VOLTAGE MOD.
38
AUXILLIARY MODULE
POWER SUPPLY MODULE
20

FIG 4

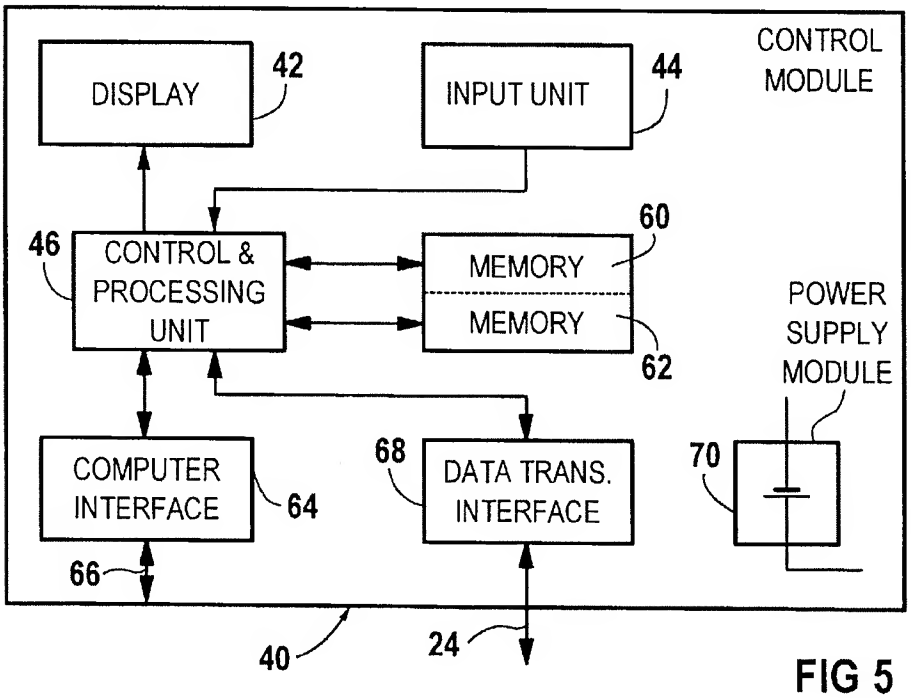

CONTROL MODULE
DISPLAY
42
INPUT UNIT
44
46
CONTROL & PROCESSING UNIT
60
MEMORY
MEMORY
62
POWER SUPPLY MODULE
COMPUTER INTERFACE
64
68
DATA TRANS. INTERFACE
70
66
40
24

FIG 5

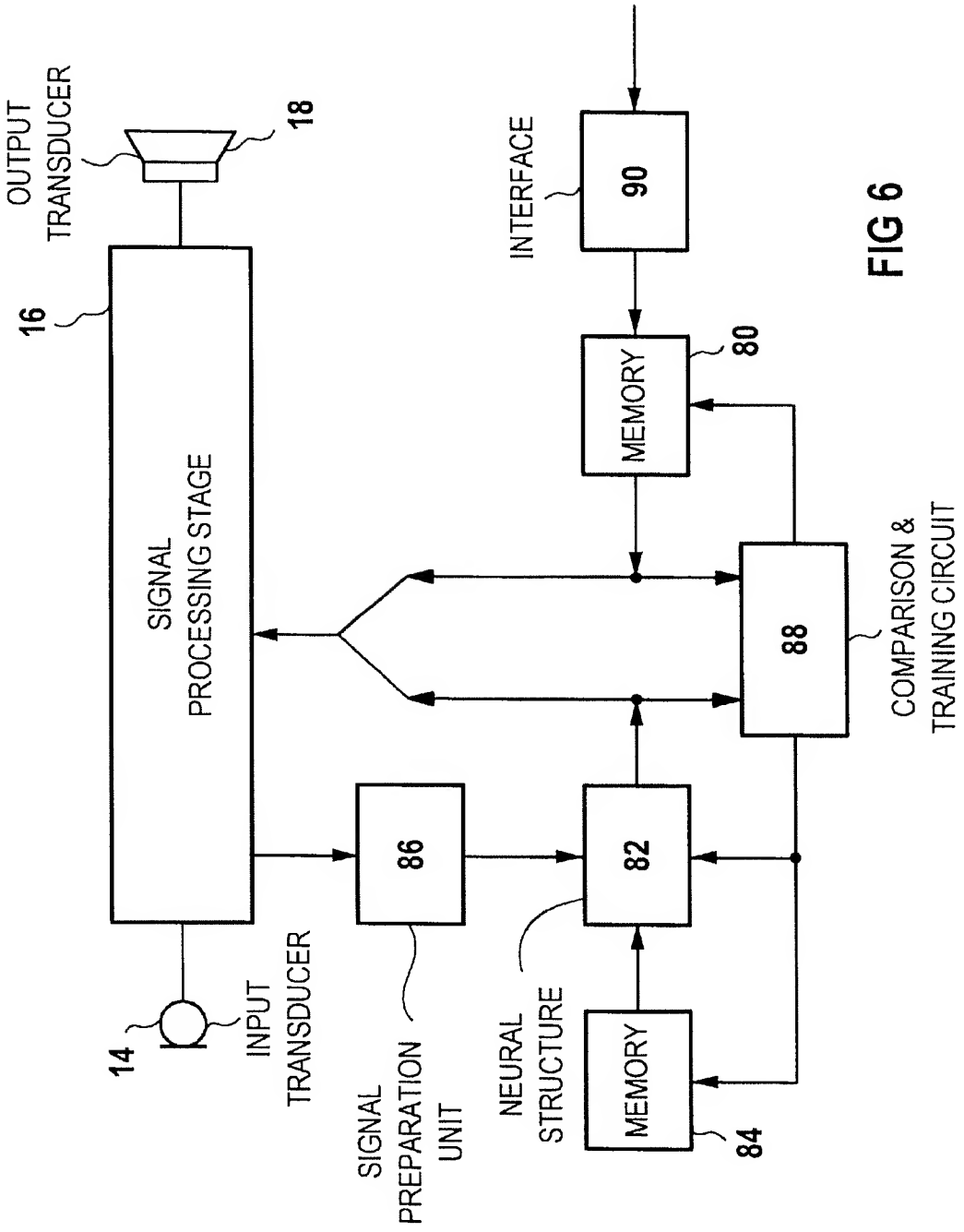

OUTPUT TRANSDUCER
18
16
SIGNAL PROCESSING STAGE
14
INPUT TRANSDUCER
SIGNAL PREPARATION UNIT
86
NEURAL STRUCTURE
82
MEMORY
84
INTERFACE
90
MEMORY
80
88
COMPARISON & TRAINING CIRCUIT

FIG 6

PROGRAMMABLE HEARING AID SYSTEM AND METHOD FOR DETERMINING OPTIMUM PARAMETER SETS IN A HEARING AID

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention is directed to programmable hearing aid system, as well as a method for determining optimum parameter sets in a hearing aid.

2. Description of the Prior Art

In a programmable hearing aid several parameter sets are generally stored so as to be selectable by the user. These parameter sets being known as hearing programs. Each of these parameter sets represents the settings, cooperatively matched to one another, of all signal processing parameters for a particular acoustic hearing or environmental situation (e.g. an environmental situation "quiet," i.e. without disturbing background noise, or an environmental situation with low-frequency disturbing noise, etc.). The wearer of the hearing aid can select the suitable hearing program.

A programmable hearing aid of this sort is known from Europearn Application 0 064 042. This hearing aid has a microphone, an earphone, a signal processor and a parameter memory. Up to eight parameter sets can be written into the memory by means of an external programming unit. By the actuation of a switch, the stored parameter sets are called one after the other and are supplied to the signal processor. The user can thus match the signal transmission function of the signal processor optimally to the current hearing situation.

In this known hearing aid system, the parameter set allocated to each hearing situation is determined during the adaptation of the hearing aid, i.e. by a hearing aid acoustician. It is difficult, however, to determine the optimal parameter set for different acoustic environmental situations of the hearing-impaired person in this manner, since the actual acoustic characteristic quantities thereof are finally dependent on individual data. For example, if a hearing aid wearer requires an "in the car" hearing program, because that person often travels in his or her own car, an optimal setting of the parameters for this program must be based on the acoustic characteristic quantities of that car, which in turn depend strongly on the type of car and other factors.

In order to avoid the complicated determination of a suitable parameter set by the hearing aid acoustician, in the hearing aid system disclosed in European Application 0 453 450 an external control apparatus is provided that calculates signal processing parameters to be set from audiometric data, in a complicated method, and calculates characteristic data from the environmental situation. This method is costly, however, and does not always produce an optimal parameter set.

An additional difficulty in the two above-cited methods for determining parameter sets is that even for identical hearing impairment (determined using a sound threshold audiogram), the subjective sensations of different hearing aid wearers can be different in identical acoustic environmental conditions, necessitating different optimal parameter sets for the respective wearers.

SUMMARY OF THE INVENTION

It is an object of the present invention to avoid the above problems associated with known hearing aids and hearing aid systems, and in particular to simplify, or often in practice

to enable for the first time, the determination of parameter sets that are individually optimally matched to different hearing situations in a hearing aid.

The above object is achieved in accordance with the principles of the present invention in a hearing aid system having a programmable hearing aid with a signal transmission path therein including a signal processor which sets transmission characteristics in the signal path dependent on a stored parameter set. The parameter set is stored in a memory in the programmable hearing aid, and the system also includes matching means, having access to the memory in the hearing aid, for allocating respective parameter sets to different hearing situations. The memory means includes a first memory for storing a number of different parameter sets, input means for identifying a number of different hearing situations and for allowing a wearer of the hearing aid to select and allocate a parameter set for each hearing situation each time it occurs. In a training phase, the hearing situations may each arise at a number of different times, and each time the wearer of the hearing aid makes an allocation of a parameter set to the current hearing situation. These allocations, produced over time during the training phase, are stored in a second memory. A control and processing means evaluates the allocations in the second memory, such as based on their frequency, for assigning a parameter set to each hearing situation dependent on these allocations. For example, for each hearing situation, the control and processing means can identify the parameter set most frequently selected by the user as being appropriate for that hearing situation, and the control and processing means then permanently allocates that parameter set to that hearing situation in the parameter set memory in the programmable hearing aid. This configures the hearing aid so that, in the future, each time that hearing situation arises, the hearing aid will identify the hearing situation and select the allocated parameter set for use in setting the transmission characteristics as long as that hearing situation prevails.

Since the parameter set memory in the programmable hearing aid can be reprogrammed, i.e., the contents thereof can be altered, for example, if the users hearing impairment changes, the term "permanently stored" as used in the context of this parameter set memory means that the allocations of the respective parameter sets are stored in the parameter set memory so as to be unchanged unless and until a reprogramming takes place. The term "permanently stored", therefore, does not mean that the parameter set allocations are forever unalterable.

The above object is also achieved in a method for determining an optimal parameter set for controlling the transmission characteristics of a programmable hearing aid in each of a number of different hearing situations, wherein a user wearing the hearing aid, in a training phase, experiences a number of different hearing situations occurring at different times, and for each hearing situation, the wearer of the hearing aid selects one of a number of different trial parameter sets for use in that hearing situation, each time the hearing situation occurs. In this training phase, the allocations of the different trial parameter sets to the different hearing situations are stored, and after completion of the training phase, these allocations are evaluated to permanently assign one of the trial parameter sets to each hearing situation. The permanent assignment can be, for example, on the basis of the frequency during the training phase by which the hearing aid user selected a particular trial parameter for a particular hearing situation. In a configuration phase, the parameter set memory in the hearing aid is then configured (programmed) based on the evaluation of the allocations so

as to permanently store one parameter set for each hearing situation. In the future operation of the hearing aid, when a particular hearing situation arises, the parameter set allocated thereto as being optimum when then be retrieved from the memory in the hearing aid, and used to set the transmission characteristics of the hearing aid, as long as that particular hearing situation prevails.

The programmable hearing aid can "identify" which of the different hearing situations is currently in existence either by the user identifying that hearing situation, such as by a switch or by a remote control, or the programmable hearing aid can include a trainable network, such as a neural structure, which can, over time, "learn" when a particular hearing situation is present. The identification of the current hearing situation is then undertaken fully automatically within the hearing aid itself, without any necessity of intervention by the hearing aid wearer.

The invention is based on the fundamental concept of not attempting to generate a predetermined parameter set allocated to each hearing situation of a programmable hearing aid during the adjustment by the hearing aid acoustician, but rather to make several trial parameter sets for each hearing situation available to the wearer at the time the wearer first uses the hearing aid. In an optimization phase, so that the hearing aid wearer can then determine which parameter set is individually best suited for him or her in various individual hearing situations. This parameter set is then finally fixedly allocated to that hearing situation.

An advantage of the inventive solution is that the matching of the hearing aid to the various hearing situations is better achieved with conventional procedures, since it is individually oriented according to the real acoustic environmental conditions of the personal life situations of the hearing-impaired person. Moreover, the matching can largely be carried out by the hearing aid wearer, so that it is less costly.

In different embodiments of the invention, summarized below, the main functions of the matching means are differently distributed to different modules.

In one embodiment at least the first memory for the trial parameter sets, the second memory for the allocations decided on by the user, and the control and processing unit are provided in an external control module that is connected wirelessly with a mobile auxiliary module. The latter contains a receiver that receives data from the control module and forwards it to the hearing aid.

In a second embodiment the modules identified in the first embodiment are contained in the mobile auxiliary module, while the external control module essentially contains only operational elements (i.e., input keys and a display), as well as one or several interfaces.

In a third embodiment, the modules can be grouped as in either the first or second embodiments but the auxiliary module is omitted, and its functions are permanently integrated into the hearing aid.

A fourth embodiment is constructed as described for the third embodiment, but after the termination of the matching phase, the control module serves as a normal remote control of the hearing aid. The matching functions are then deactivated.

In a fifth embodiment all modules of the matching means, including the operational elements, are integrated into the mobile auxiliary module, to be worn on the body. The control module can be omitted.

The evaluation of the allocation data stored in the second memory of the matching means during the matching phase preferably ensues either in an external evaluation computer or in the control module. Besides the evaluation, a constant monitoring of the allocation data also can take place only at the end of the matching phase, e.g. in order to determine whether no optimal trial parameter set is present for a hearing situation, and the hearing aid acoustician must thus be consulted to program new trial parameter sets. In an alternative embodiment, the matching means produces new parameter sets according to predetermined rules.

In a preferred embodiment, the hearing aid has a neural structure and a comparison and training circuit. The neural structure continuously evaluates acoustic input signals. The comparison and training circuit makes it possible to train the neural structure according to the parameter sets selected for each hearing situation during a training phase. After the conclusion of the training phase, the neural structure independently determines matching signal processing parameters from the input signals, so that the hearing aid user never again has to indicate the currently present hearing situation.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** illustrates the arrangement of the components of an inventive hearing aid system in an embodiment with a behind-the-ear hearing aid.

FIG. **2** illustrates the arrangement of the components of an inventive hearing aid system in an embodiment with an in-the-ear hearing aid.

FIG. **3** shows a view of an external control module used in the inventive system and method.

FIG. **4** shows a block diagram of an auxiliary module used in the inventive system and method.

FIG. **5** shows a block diagram of the external control module used in the inventive system and method.

FIG. **6** shows a block diagram of a hearing aid with a neural structure used in the inventive system and method.

DESCRIPTION OF THE PREFERRED EMBODIMENT

In FIG. **1**, the outline of an ear is shown as a dotted line, with a hearing aid **10** to be worn behind the ear, on which an auxiliary module **20** is detachably plugged. The hearing aid **10** and the auxiliary module **20** are electrically connected with one another via contact surfaces. Via this connection, parameter sets can be programmed into the hearing aid **10**, which sets determine the signal processing characteristic in the hearing aid **10**. The auxiliary module **20** permits the exchange of data with an external control module **40** via a wireless data transmission path **24**.

As a modification of the hearing aid system shown in FIG. **1**, FIG. **2** shows a hearing aid **10'** to be worn in the ear that is connected with the auxiliary module **20**, to be worn behind the ear, via a connection line **12**. The connection line **12** is detachably connected to the hearing aid **10'** by means of known connection elements (plugs/sockets, etc.), as are used, for example, for the hard-wired programming of hearing aids.

FIG. **3** shows details of the operating and display elements of the external control module **40**, constructed in a manner similar to a remote control for electronic entertainment systems. A display **42**, constructed for example as an alphanumeric LCD display, serves for user control. For example, the set hearing situation can be displayed in a first line (such as hearing situation **53**: workplace in the example of FIG. **3**), and the respectively allocated parameter set can be displayed

in a second line (such as parameter set E2 in the example). Other texts that were programmed in during the programming of the control module **40** can also be displayed. An input unit **44**, constructed as a keyboard or keypad, has several keys or pads, in particular keys **48** for setting the hearing situation, keys **50** for the allocation of a parameter set to the hearing situation, a key **52** for confirmation and a key **54** for the correction of erroneous inputs. In order to organize the operation of the control module **40** simply, only a few clearly identified keys without double functions are provided; for example, keys for a maximum of four hearing situations, respectively with a maximum of four trial parameter sets, in the control module **40** shown in FIG. **3**.

The auxiliary module **20** shown in FIG. **4** has an interface **22** for bidirectional data transmission to the hearing aid **10** (or **10'**) via contacts or via the electrical connection line **12**. A data transmission interface **26**, formed by an infrared light-emitting diode and a photosensor, serves to provide the bidirectional data transmission path **24** to the external control module **40**. The data transmission path **24** is preferably wireless. Visible or infrared light, radio-frequency broadcast waves, ultrasound, electrical induction, etc., can be employed. The interfaces **22** and/or **26** can also be constructed more simply as unidirectional interfaces that enable transmission of parameter sets only in the direction to the hearing aid **10** or **10'**.

The interfaces **22** and **26** are connected with one another, as well as with a control and processing unit **30**. The latter enables access to several read-only memories and/or write/read memories, in particular to a first memory **32** for the trial parameter sets and to a second memory **34** for the allocations chosen by the user during the optimization phase. In addition, a module **36** for the production of a possibly required programming voltage for the hearing aid **10** or **10'**, as well as a power supply module **38**, are provided. The module **36** is connected to the connection line **12**, and is controlled by the control and processing unit **30**. The power supply module **38** supplies all the named components, and furthermore is connected with the hearing aid **10** or **10'** via the connection line **12**.

The auxiliary module **20** is shown in FIG. **4** in an embodiment with complete functionality. In other embodiments, in which some functions are for example, taken over by the control module **40**, some modules can be omitted. For example, the first and second memories **32** and **34** need only be provided either in the auxiliary module **20** or in the control module **40**. The control and processing unit **30** can then be constructed more simply, or even can be omitted entirely.

FIG. **5** shows the construction of the external control module **40**. The display **42** and input unit **44**, already described in connection with FIG. **3**, are connected with a control and processing unit **46**, to which are connected first and second memories **60** and **62**, a computer interface **64** and a data transmission interface **68** to the auxiliary module **20**. In addition, a power supply module **70** is provided for the named modules. The computer interface **64** is connected with a terminal **66** for an external evaluation computer. Via the computer interface **64**, on the one hand trial parameter sets can be transmitted from the evaluation computer to the control module **40** before the beginning of the optimization phase, and, on the other hand, allocation data can be transmitted from the control module **40** to the evaluation computer after the termination of the optimization phase.

The control module **40** is also shown in FIG. **5** in an embodiment with complete functionality. According to the distribution of the functions of the matching means among the auxiliary module **20** and to the control module **40**, individual modules can be omitted or can be simplified. The computer interface **64** can be omitted if the entry of the trial parameter sets ensues via the input unit **44**, and the evaluation of the allocation data is carried out by the control and processing unit **46**. Moreover, the control and processing unit **46** for generating new or modified trial parameter sets can be set up according to rules that are programmed in or that are fixedly predetermined.

FIG. **6** shows the circuit of a complexly constructed hearing aid **10** or **10'**, specified in more detail below. For the previously described embodiments of the hearing aid system, a hearing aid **10** or **10'** is sufficient, in which, of the components shown in FIG. **6**, there are provided only an input transducer **14** constructed as a microphone, an output transducer **18** constructed as an earphone, a signal processing stage **16** with a transmission characteristic determined by the aforementioned parameters of a parameter set, a memory **80** for at least one parameter set of the signal processing stage **16**, and an interface **90** to the matching means. In an embodiment, the interface **90** is connected with the auxiliary module **20** via the electrical connection line **12**.

For the configuring and optimization of the parameters of the hearing aid, according to an exemplary embodiment of the inventive method the hearing aid acoustician first determines the hearing situations for which the wearer of the hearing aid wishes to individually determine the parameter sets (also called hearing programs). Examples of hearing situations might include: "at work," "conversing in the car," "listening to music at home," etc. For each of these hearing situations, several trial parameter sets are determined, dependent on the hearing impairment of the wearer of the hearing aid using matching software that runs on the external evaluation computer. The determined parameter sets are transmitted to the control module **40** via the computer interface **64**, and are either stored there in the first memory **60** or are transmitted further via the data transmission path **24** to the auxiliary module **20**, and are stored in the first memory **32** thereof.

For the parameter optimization phase, the control module **40** and the mobile auxiliary module **20** are provided to the hearing-impaired person. If the hearingimpaired person is in a hearing situation typical for him or her, he or she can first select the hearing situation via the control module by means of the keys **48**, and can subsequently respectively activate one of the trial parameter sets allocated thereto by means of the keys **50**. This set is now transmitted from the control module **40** to the mobile auxiliary module **20**, is programmed into the hearing aid **10** or **10'** by this module, and is activated there. If the hearing-impaired person has found the optimal set of parameters for the selected hearing situation, he or she can store it by actuating the confirmation key **52**. That is, it is noted in the second memory **62** of the control module **40** (in the second memory **34** of the auxiliary module **20**) that an allocation of this parameter set to the identified hearing situation has taken place.

After the user's optimization phase is completed, the second memory **62** of the control module **40** (the second memory **34** of the auxiliary module **20**) is read out by the hearing aid acoustician, and it is determined the frequency with which allocation of hearing situations to parameter sets has been made. The parameter set with the most frequent allocation for a particular hearing situation is stored as the corresponding hearing program in the hearing aid **10** or **10'** for that hearing situation. This is done for each hearing situation.

The optimization phase is terminated, and it remains only for the user to wear the hearing aid **10** or **10'** (and no longer the matching means including the auxiliary module **20** and the control module **40**). If the hearing aid system is designed so that the control module **40** communicates directly with the hearing aid **10** or **10'**, the control module **40** can then also serve as a normal remote control of the hearing aid **10** or **10'** after the end of the optimization phase. The matching functions are then deactivated. In this version, the parameter sets determined in the optimization phase can remain stored in the control module **40**, which now acts as a remote control. Only the currently desired parameter set needs to be transmitted to the hearing aid **10** or **10'**, so that the latter need have only a memory **80** for a single parameter set.

If, upon completion of the user's optimization phase the allocation frequency of some or all of the parameter sets is too low to allow the acoustician to confidently assign significance for a hearing situation, the corresponding parameter sets can be modified by the acoustician using the matching software, and can be stored again in the control module **40**. The optimal allocation can then be determined again in a second optimization phase.

In an alternative embodiment of the inventive method, the evaluation of the allocations of hearing situations to trial parameter sets ensues already during the optimization phase in the control module **40**. A too low frequency of the allocations of trial parameters to a particular hearing situation is interpreted to mean that no optimal parameter set is present for this hearing situation. The wearer of the hearing aid is then requested via the display **42** to consult his or her hearing aid acoustician, in order to have new trial parameter sets programmed in. Alternatively, these new trial parameter sets can be generated in the control module **40** according to fixedly predetermined rules or rules that can be programmed in.

In another variant embodiment of the invention, the hearing aid **10** or **10'** is constructed according to FIG. **6**. Besides the components already specified above, this hearing aid **10** or **10'** has a neural structure **82**, also called a neural network, a memory **84** for parameters of the neural structure **82**, a signal preparation unit **86** and a comparison and training circuit **88**. The signal preparation unit **86** is connected with the signal processing stage **16** at a suitable top point, and supplies suitably prepared signals to the neural structure **82**, which correspond to the items of acoustic information received by the input transducer **14**.

The memory **84** contains parameters that control the output behavior of the neural structure **82**. The memory **84** is connected with the neural structure **82**, as well as with the comparison and training circuit **88**. The comparison and training circuit **88** controls the neural structure **82**, the memory **84** for the neural structure **82** and the memory **80** for parameter sets. The outputs of the memory **80** or of the neural structure **82** are connected with the comparison and training circuit **88**, as well as with a parameter input of the signal processing stage **16**, via which the transmission characteristic of the signal processing stage **16** can be set. By means of the comparison and training circuit **88**, it is determined among other things whether the outputs of the neural structure **82**, the parameters stored in the memory **80** or a mixture of the two are used to control the signal processing stage **16**.

From European Application 0 712 263, a hearing aid **10** or **10'** is known in which the parameters controlling the signal processing are determined by a neural structure. The content of European Application 0 712 263 is incorporated herein by reference, in particular with respect to the construction of the signal preparation unit **86** (see FIG. **3** of European Application 0 712 263, with the associated specification) and the neural structure **82** (see FIG. **4** to FIG. **8** of European Application 0 712 263, with the associated specification). European Application 0 712 263 does not, however, describe how the training of the neural structure **82** can take place.

According to the inventive system and method, trial parameter sets are first determined for the training of the neural structure **82**, and thus for the programming of the hearing aid system. During the optimization phase, the user first communicates the parameter set believed to be optimal for the momentary hearing situation to the hearing aid **10** or **10'**, via the interface **90** in the way specified above. This is written into the memory **80**. Independently of this, the neural structure **82** calculates a proposed parameter set from the data originating from the signal preparation unit **86**.

During the optimization phase, the comparison and training circuit **88** continuously compares the parameter set believed to be optimal by the user and written into the memory **80**, with the parameter set determined by the neural structure **82**. An error identifier is obtained from the deviations of these parameter sets according to a predetermined algorithm (e.g. a learning algorithm for neural networks according to the prior art). Based on this error identifier, the comparison and training circuit modifies the parameters, contained in the memory **84**, for the neural structure **82**. In this way, the neural structure **82** is trained during the optimization phase until it can by itself determine suitable parameter sets for each environmental acoustic condition, as it arises, with satisfactory precision.

At the beginning of the optimization phase (training phase), the signal processing stage **16** receives its control parameters exclusively from the memory **80** for the parameter set entered by the user; as the training success progresses, these parameters are increasingly taken from the neural structure **82**. After the termination of the training phase, the signal processing stage **16** continues to receive its control parameters only from the neural structure **82**. The matching means is then no longer needed by the hearing aid wearer.

Although the present invention has been described with reference to a specific embodiment, those of skill in the art will recognize that changes may be made thereto without departing from the scope and spirit of the invention as set forth in the appended claims.

We claim as our invention:

1. A hearing aid system comprising:

a programmable hearing aid having a housing adapted to be worn at an ear, said housing containing an input transducer and an output transducer with a signal path therebetween, signal processing means connected in said signal path for influencing a signal in said signal path dependent on a parameter set, and a parameter set memory accessible by said signal processor means for storing at least one parameter set for use by said signal processing means;

identifier means in said housing for identifying a current hearing situation defining an environment in which said programmable hearing aid is disposed;

matching means for allocating respective parameter sets in a plurality of parameter sets to different hearing situations, said matching means including a first memory for storing said plurality of parameter sets, input means for identifying a current hearing situation

among a succession of hearing situations and for allowing a wearer of said hearing aid to select and allocate a parameter set, for said plurality of parameter sets, for each hearing situation each time it occurs, a second memory for storing respective allocations made by said user among said parameter sets and said hearing situations, and control and processing means for evaluating said allocations in said second memory for assigning a parameter set among said plurality of parameter sets to each hearing situation dependent on said allocations and for programming said parameter set memory with said parameter sets respectively allocated to said hearing situations, said parameter set memory then supplying to said hearing situations, said parameter set memory then supplying to said signal processing means, when a current hearing situation is identified by said identifier means, the parameter set allocated to the current hearing situation; and

said matching means comprising an external control module. including at least said input means, and an auxiliary module, said auxiliary module containing a remainder of said matching means not contained in said external control module, and means for wirelessly transmitting data at least from said external control module to said auxiliary module, said auxiliary module being temporarily mechanically connectable to said housing and adapted to be worn at an ear together with said housing during a matching procedure, consisting of a training phase and a hearing aid configuration phase, in which said respective parameter sets in said plurality of parameter sets are allocated to different hearing situations and being electrically connected to said signal processing means, and after said matching procedure said auxiliary module being removable from said housing.

2. A hearing aid system as claimed in claim **1** wherein said auxiliary module contains an interface to said programmable hearing aid and a data transmission interface to said external control module, and wherein said external control module contains, in addition to said input means, said first and second memories, said control and processing means, and a data transmission interface to said auxiliary module.

3. A hearing aid system as claimed in claim **1** wherein said auxiliary module contains an interface to said hearing aid, said first and second memories, said control and processing means, and a data transmission interface to said external control module, and wherein said external control module, in addition to said input means, contains a data transmission interface to said auxiliary module.

4. A hearing aid system as claimed in claim **1** further comprising a remote control means for operating said programmable hearing aid, including said identifier means, said remote control means containing said external control module.

5. A hearing aid system as claimed in claim **1** wherein said matching means comprises an external control module, including at least said input means, and an auxiliary module contained in said programmable hearing aid, said auxiliary module containing a remainder of said matching means not contained in said external control module, and means for wirelessly transmitting data at least from said external control module to said auxiliary module.

6. A hearing aid system as claimed in claim **5** wherein said auxiliary module contains an interface to said programmable hearing aid and a data transmission interface to said external control module, and wherein said external control module contains, in addition to said input means, said first and second memories, said control and processing means, and a data transmission interface to said auxiliary module.

7. A hearing aid system as claimed in claim **5** wherein said auxiliary module contains an interface to said hearing aid, said first and second memories, said control and processing means, and a data transmission interface to said external control module, and wherein said external control module, in addition to said input means, contains a data transmission interface to said auxiliary module.

8. A hearing aid system as claimed in claim **5** further comprising a remote control means for operating said programmable hearing aid, including said identifier means, said remote control means containing said external control module.

9. A hearing aid system as claimed in claim **1** wherein said matching means includes display means for displaying an alphanumeric indication of said parameter sets and said different hearing situations.

10. A hearing aid system as claimed in claim **1** further comprising means for determining an optimal allocation of each parameter set to each hearing situation from said allocations stored in said second memory.

11. A hearing aid system as claimed in claim **10** wherein said means for determining an optimal allocation includes means for determining whether an optimal allocation of a respective parameter set to each hearing situation can be determined from said allocations stored in said second memory.

12. A hearing aid system as claimed in claim **1** further comprising a neural structure, a neural structure memory for storing parameters for said neural structure, and comparison and training means for training said neural structure according to said parameter sets respectively allocated to said hearing situations, by modifying said parameters in said neural structure memory.

13. A method for determining an optimal parameter set for controlling transmission characteristics of a programmable hearing, aid having a housing containing a signal processor, in each of a plurality of different hearing situations, comprising the steps of:

temporarily mechanically connecting an auxiliary module to said housing and temporarily electrically connecting said auxiliary module to said signal processor: providing a remote control in a wireless communication with said auxiliary module;

in a training phase, wearing said hearing aid and said auxiliary module by a user in said plurality of different hearing situations;

in said training phase, making a plurality of different trial parameter sets from said auxiliary module available for selection by said user in each hearing situation each time a hearing situation each time a hearing situation occurs;

in said training phase, for each hearing situation, said user selecting one of said trial parameter sets deemed optimal by said user and storing an allocation in said auxiliary module of said one of said trial parameter sets to the hearing situation for which it was deemed optimal;

in a hearing aid configuration phase, evaluating all of the stored allocations of said trial parameter sets to the different hearing situations and remotely programming said hearing aid via said auxiliary module, using said remote control to assign one trial parameter set in said hearing aid to each hearing situation for controlling

said transmission characteristics of said hearing aid when said hearing situations respectively occur; and

removing said auxiliary module from said housing.

14. A method as claimed in claim **13** wherein the step of evaluating all of said stored allocations comprises evaluating a frequency for which said user selected each trial parameter set for each hearing situation.

15. A method as claimed in claim **13** comprising the additional step of identifying an allocation frequency of a parameter set to a hearing situation which is too low to be significant, and providing a message to said user of said hearing aid to select a different trial parameter set.

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(54) UPNP ENABLING DEVICE FOR HETEROGENEOUS NETWORKS OF SLAVE DEVICES

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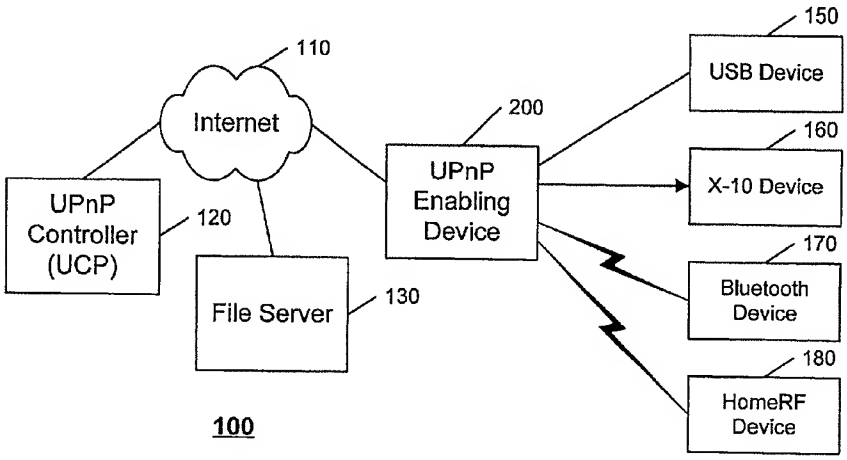
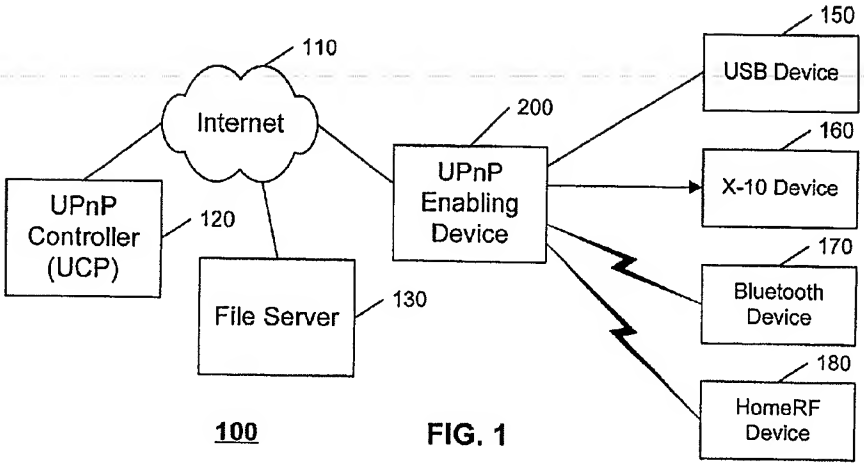
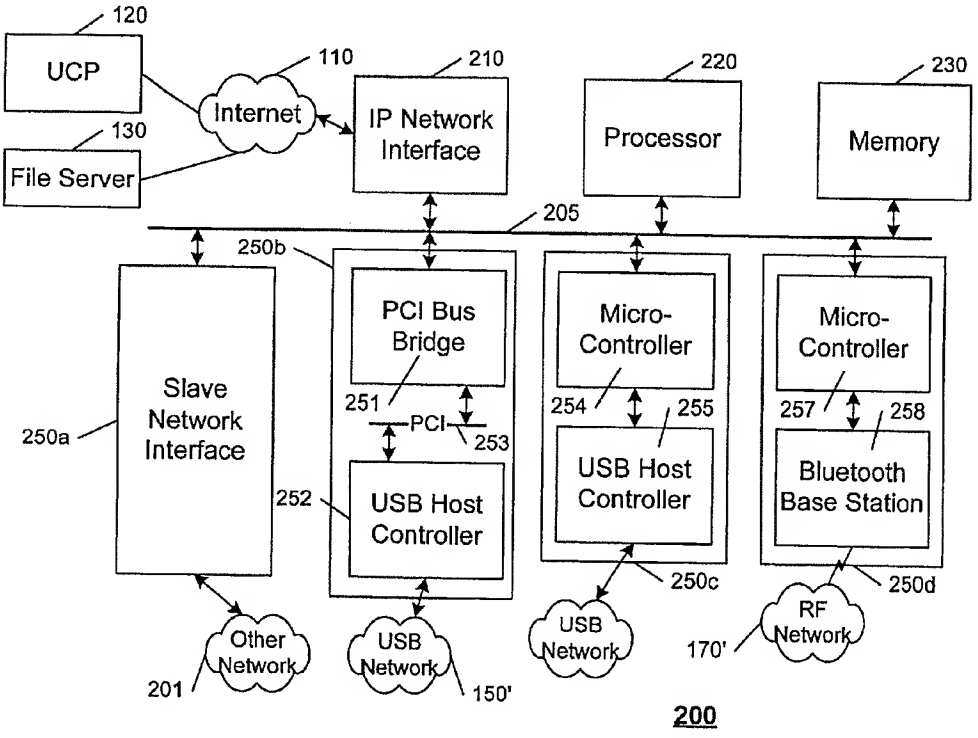
(57) ABSTRACT

A bridging device couples an IP (Internet Protocol) network to one or more non-IP networks, in order to facilitate the control of non-UPnP (Universal Plug and Play) devices by a UPnP controller on the IP network. Each of the non-IP slave networks may employ different network technologies, such as USB, Bluetooth, HAVi, Home API, HomeRF, X-10, Jini, and so on. The bridging device includes an IP network interface for receiving commands and requests from the UPnP controller, and one or more slave network interfaces that transform the received commands and requests into device and network specific commands and requests. These device and network specific commands and requests are communicated to the controlled non-UPNP device, via the slave network, using the slave network's protocol. The bridging device also communicates event status messages to the UPnP controller, corresponding to the non-UPnP devices' response to the UPnP controller's commands and requests. The bridging device also includes enabling logic to support the UPNP addressing, discovery, and description processes for each of the devices on the non-IP network. To minimize the storage requirements at the bridging device, the bridging device is configure to use a file server that is also resident on an IP network, wherein the file server contains the detailed information required to effect the appropriate UPnP addressing, discovery, and description processes, and other information-laden tasks, as required.





100 FIG. 1



200

FIG. 2

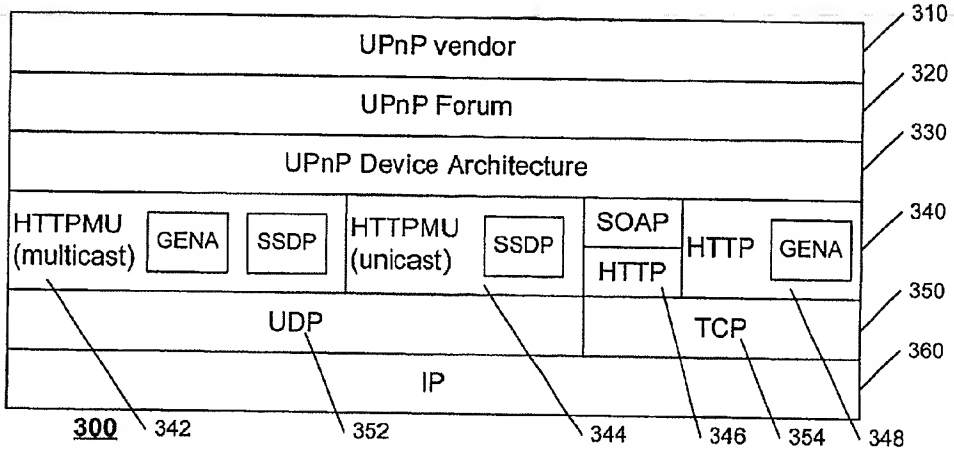


FIG. 3 [Prior Art]

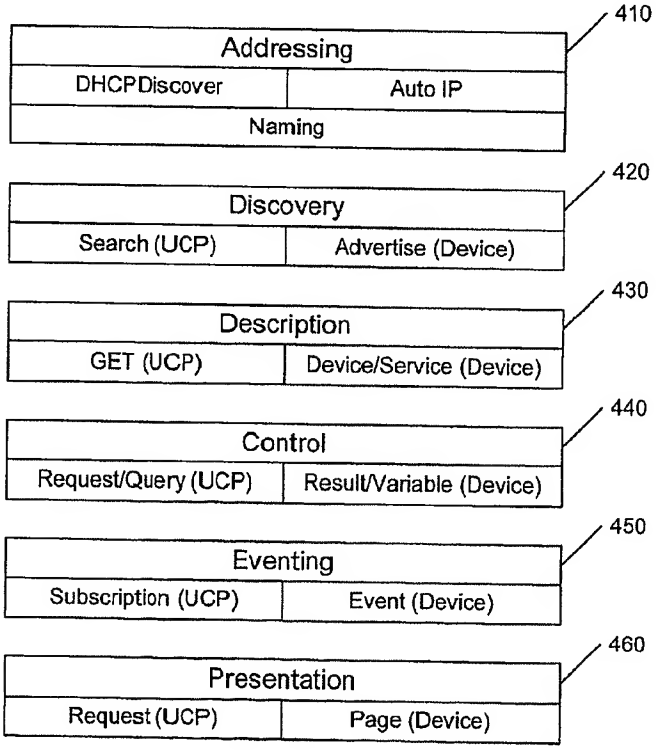


FIG. 4

[Prior Art]

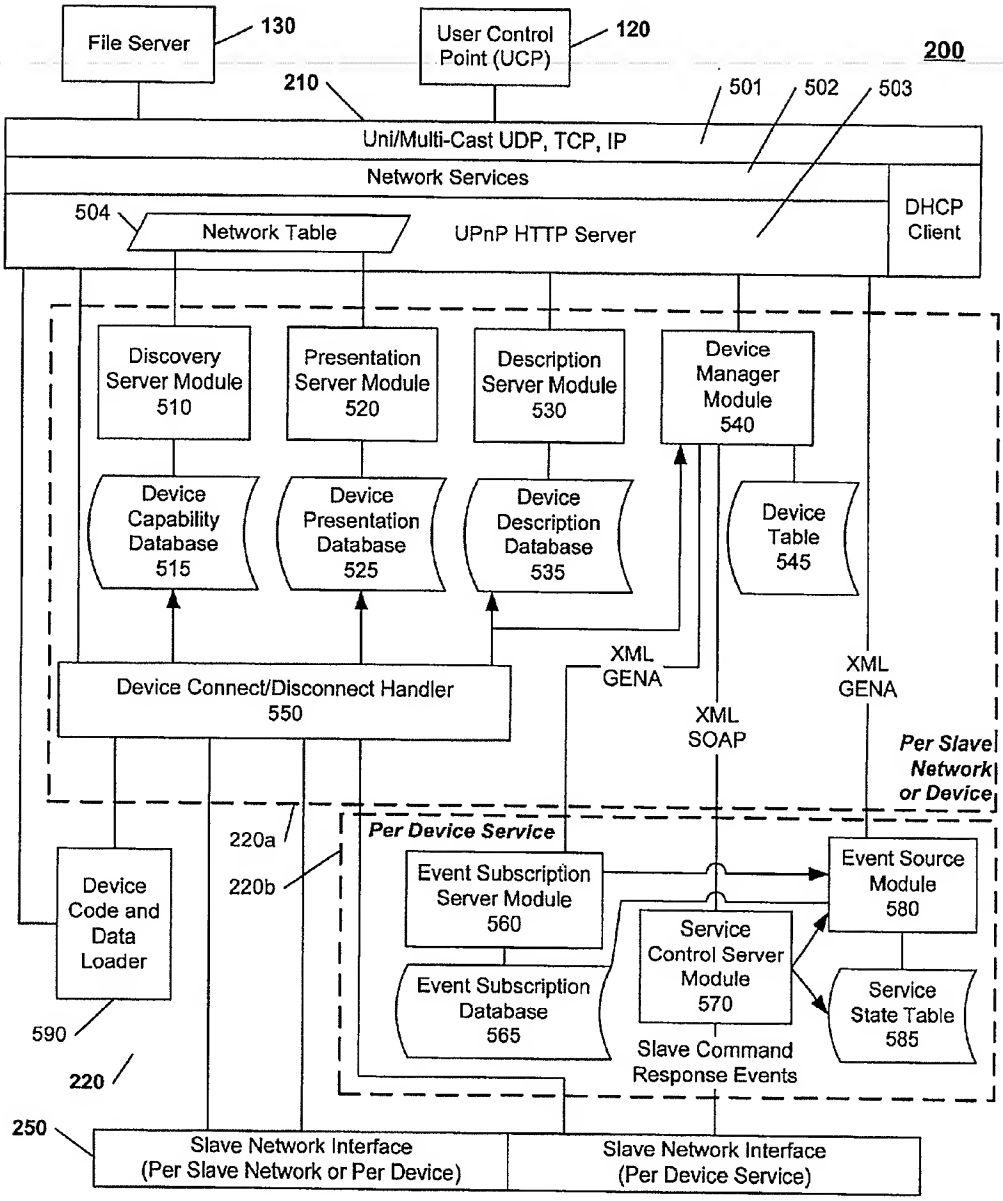

File Server
130
User Control Point (UCP)
120
200
210
501
502
503
Uni/Multi-Cast UDP, TCP, IP
Network Services
504
Network Table
UPnP HTTP Server
DHCP Client
Discovery Server Module 510
Presentation Server Module 520
Description Server Module 530
Device Manager Module 540
Device Capability Database 515
Device Presentation Database 525
Device Description Database 535
Device Table 545
XML GENA
XML SOAP
XML GENA
Device Connect/Disconnect Handler 550
Per Slave Network or Device
Per Device Service
220a
220b
Device Code and Data Loader
Event Subscription Server Module 560
Event Source Module 580
Service Control Server Module 570
Service State Table 585
Event Subscription Database 565
Slave Command Response Events
590
220
250
Slave Network Interface (Per Slave Network or Per Device)
Slave Network Interface (Per Device Service)

FIG. 5

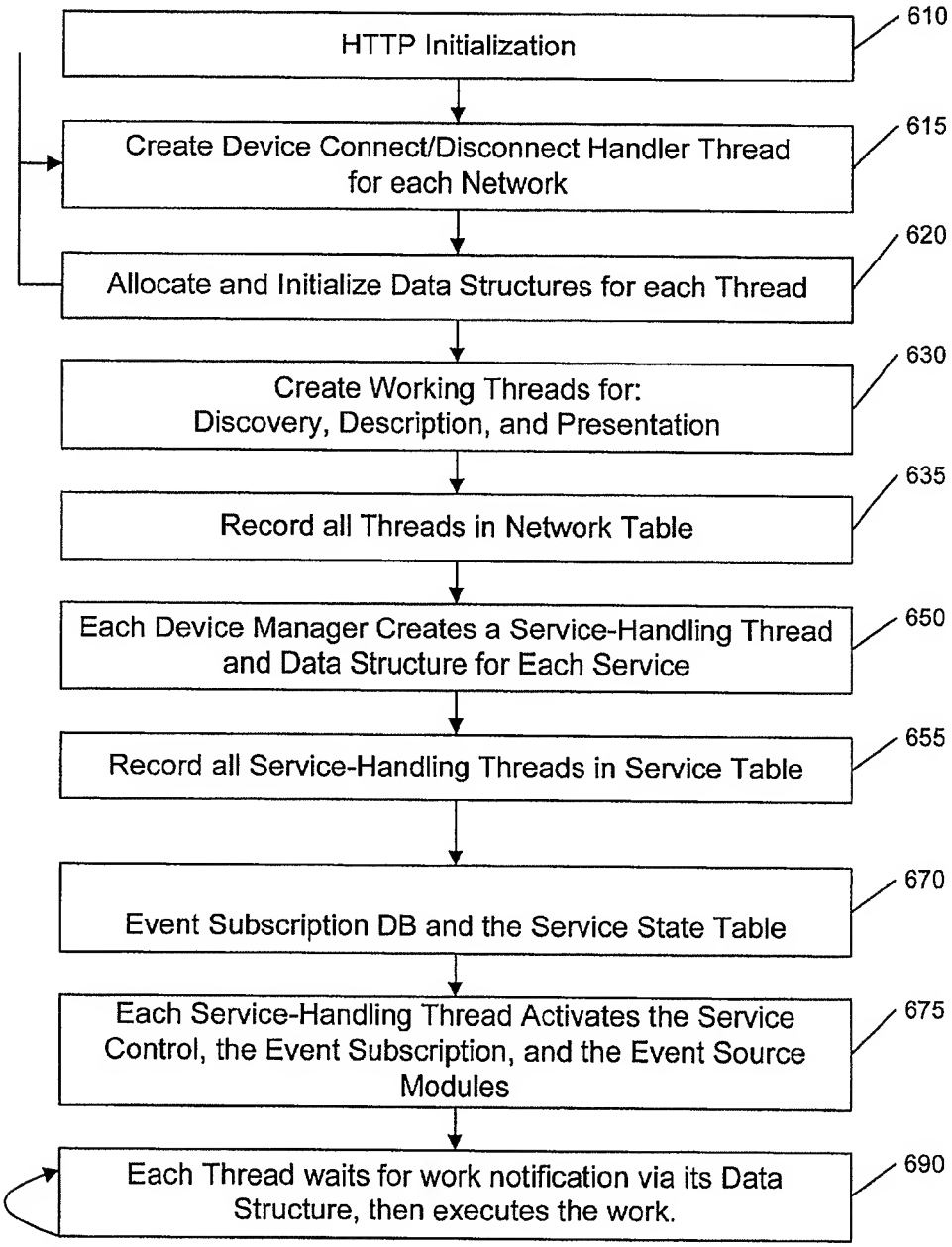

HTTP Initialization
610
Create Device Connect/Disconnect Handler Thread for each Network
615
Allocate and Initialize Data Structures for each Thread
620
Create Working Threads for: Discovery, Description, and Presentation
630
Record all Threads in Network Table
635
Each Device Manager Creates a Service-Handling Thread and Data Structure for Each Service
650
Record all Service-Handling Threads in Service Table
655
Event Subscription DB and the Service State Table
670
Each Service-Handling Thread Activates the Service Control, the Event Subscription, and the Event Source Modules
675
Each Thread waits for work notification via its Data Structure, then executes the work.
690

FIG. 6

UPNP ENABLING DEVICE FOR HETEROGENEOUS NETWORKS OF SLAVE DEVICES

BACKGROUND OF THE INVENTION

[0001] 1. Field of the Invention

[0002] This invention relates to the field of control systems, and in particular to the control of non-UPnP-compliant slave devices via a Universal Plug and Play (UPnP) object, or application.

[0003] 2. Description of Related Art

[0004] "Universal Plug and Play (UPnP) is an architecture for pervasive peer-to-peer network connectivity of intelligent appliances, wireless devices, and PCs of all form factors. It is designed to bring easy-to-use, flexible, standards-based connectivity to ad-hoc or unmanaged networks whether in the home, in a small business, public spaces, or attached to the Internet. Universal Plug and Play is a distributed, open networking architecture that leverages TCP/IP and the Web technologies to enable seamless proximity networking in addition to control and data transfer among networked devices in the home, office, and public spaces."[1]

1 "Universal Plug and Play Device Architecture", Version 1.0, Jun. 8, 2000, © 1999-2000 Microsoft Corporation, incorporated by reference herein.

[0005] Other networking solutions are also available for control and data transfer among networked devices in the home, office, and public spaces. Standards continue to be developed which will allow devices of varying types and varying vendors to be controlled by a common controller. The HAVi architecture, the Home API initiative, the Universal Serial Bus (USB), HomeRF Lite, and the Bluetooth standard, each involving substantial contributions from Philips Electronics, the OSGI/Jini technology of Sun Microsystems, Inc., and others, have been developed to enhance the interoperability of multiple devices in a network.

[0006] Each of the available network solutions has particular advantages and disadvantages. The USB interface, for example, is relatively inexpensive, and, as such, is incorporated into many computer peripheral devices, such as keyboards, mice, pointing devices, and so on. The USB also provides a fairly high speed connectivity at this low cost, and has been adopted as a standard interface for video information transfer, such as from a video camera. The USB, however, has a limited cable length specification of less than 30 meters, and in some applications, less than 5 meters. The UPnP networking architecture, on the other hand, uses the TCP/IP protocol, which is currently used for world-wide communication networks, such as the world-wide-web. The TCP/IP, however, is a more capable, and hence more complex and costly protocol, which is typically embodied via a high speed Ethernet connection. Although TCP/IP is a viable networking solution for computers, high speed printers, servers, and the like, its inherent complexity does not encourage its use in consumer devices such as cameras, DVD players, recorders, and the like. In like manner, the Bluetooth standard supports the use of wireless devices in a networked environment, but is unsuitable for TCP/IP-based communications and control, such as provided by the UPnP standard.

[0007] The advantages and disadvantages of each networking solution are likely to result in a variety of networks being installed in a typical home or office environment. With the existence of multiple devices in a typical environment, there is an every increasing need for devices and systems that provide a bridge between and among such heterogeneous networks.

BRIEF SUMMARY OF THE INVENTION

[0008] It is an object of this invention to provide a method and system for coupling IP networks with non-IP networks. It is a further object of this invention to provide a method and system that allows for the control of non-UPnP-compliant devices from a UPnP-compliant controller. It is a further object of this invention to enable the control of non-UPnP-compliant slave devices without modification to the slave devices.

[0009] These objects and others are achieved by providing a bridging device that couples an IP network to one or more non-IP networks. Each of the non-IP networks may employ different network technologies, such as USB, Bluetooth, IEEE 1394, Home API, HomeRF, Firefly, X-10, and so on. The bridging device includes an IP network interface for receiving commands and requests from a UPnP controller on an IP network, and one or more slave network interfaces that transform the received commands and requests into device and network specific commands and requests. These device and network specific commands and requests are communicated to the controlled device, via the slave network, using the slave network's protocol. The bridging device also communicates event status messages to the UPnP controller, corresponding to the non-UPnP devices' response to the UPnP controller's commands and requests. The bridging device also includes enabling logic to support the UPnP addressing, discovery, and description processes for each of the devices on the non-IP network. To minimize the storage requirements at the bridging device, the bridging device is configured to use a file server that is also resident on the IP network, wherein the file server contains the detailed information required to effect the appropriate UPnP addressing, discovery, and description processes, and other information-laden tasks, as required.

BRIEF DESCRIPTION OF THE DRAWINGS

[0010] The invention is explained in further detail, and by way of example, with reference to the accompanying drawings wherein:

[0011] **FIG. 1** illustrates an example block diagram of a system comprising a UPnP enabling device that bridges a UPnP control device to multiple non-UPnP devices in accordance with this invention.

[0012] **FIG. 2** illustrates an example block diagram of a UPnP enabling device for bridging a UPnP controller to non-IP networks, in accordance with this invention.

[0013] **FIG. 3** illustrates an example prior art UPnP protocol stack.

[0014] **FIG. 4** illustrates an example prior art UPnP process.

[0015] **FIG. 5** illustrates a more detailed example block diagram of functions performed by a UPnP enabling device in accordance with this invention.

[0016] **FIG. 6** illustrates an example flow diagram of thread creation to provide a non-blocking architecture for communications between the UPnP controllers and the non-UPnP devices, in accordance with this invention.

[0017] Throughout the drawings, the same reference numerals indicate similar or corresponding features or functions.

DETAILED DESCRIPTION OF THE INVENTION

[0018] **FIG. 1** illustrates an example block diagram of a system **100** comprising a UPnP enabling device **200** that bridges a UPnP controller, or UPnP User Control Point (UCP) **120** to multiple non-UPnP-compliant devices **150-180** in accordance with this invention. For ease of reference, UPnP-compliant objects are referred to as UPnP objects, and devices that are not UPnP-compliant are referred to as non-UPnP devices. These non-UPnP devices include, for example, devices on a USB network, a bluetooth network, a HAVi-compatible network, such as an IEEE 1394 network, a Home API network, a HomeRF network, a Firefly network, a power line network, such as an X-10 network, and a Jini-compatible network.

[0019] The UPnP enabling device **200**, in conjunction with a file server **130**, provides the interface required to effect the control of the non-UPnP devices by the UPnP user control point **120**, by emulating each of the non-UPnP devices as a UPnP-compliant device. In operation, the UPnP user control point **120** is a conventional UPnP controller that is configured to operate in conformance with the UPnP standards and protocols, issuing commands and requests to, and receiving status reports from, UPnP-compliant devices. In like manner, the non-UPnP devices **150-180** are conventional non-UPnP devices, such as USB-compliant devices **150**, X-10-compliant devices **160**, Bluetooth-compliant devices **170**, and others **180**, that are configured to receive commands from, and send status reports to, controllers that are specific to each of these non-UPnP standards and protocols. The UPnP enabling device **200** provides the bridging interface required to effectively emulate each non-UPnP device as a UPnP-controllable device for control by the UPnP user control point, and to emulate the UPnP user control point as a non-UPnP controller that conforms to each of the standards and protocols of the non-UPnP devices **150-180**.

[0020] Depending upon packaging and marketing requirements, a UPnP enabling device **200** in accordance with this invention may include support for one or more non-UPnP interfaces. For example, a USB-specific enabling device **200** may be marketed that includes the UPnP interface for communication with the UPnP user control point, and a USB port, or pair of USB ports, for communication with USB devices on a USB network. Alternatively, an embodiment of the UPnP enabling device **200** of this invention may include multiple interface options, such as a pair of USB ports, plus an RF transceiver for communicating with a device on a Bluetooth network. Also, although most controllable devices are configured to operate in a multiple-device network, the UPnP enabling device **200** may also be configured to provide an interface for devices that operate via point-to-point control, such as an infrared interface to a printer or to a television receiver. The invention is presented herein using the paradigm of a UPnP enabling device for multiple-heterogeneous-networks, for illustrative purposes, although various alternative configurations will be obvious to one of ordinary skill in the art in view of this disclosure.

[0021] **FIG. 2** illustrates an example block diagram of a UPnP enabling device **200** for bridging a UPnP user control point **120** to non-IP networks **150'**, **170'**, **201**, in accordance with this invention. Four example non-UPnP interfaces **250***a-d*, commonly referred to hereinafter as slave network interfaces, are included in the example enabling device **200** of **FIG. 2**. Any of a variety of techniques can be used to provide these interfaces **250***a-d*. Illustrated in the interface **250***b*, for example, a PCI bus **253** is used as an intermediate bus between an internal bus **205** of the enabling device **200** and a USB network **150'**. In this manner, a conventional PCI-to-USB host controller **252** can be used to provide a USB-compliant interface to the USB network **150'**. In this example interface **250***b*, a PCI bus bridge **251** transforms data on the internal bus **205** into PCI-compliant signals, and vice-versa, for communication via the PCI bus **253**. Alternatively, as illustrated by the slave interface **250***c*, a microcontroller **254** may be provided that transforms the data to and from the internal bus **205** from and to a USB host controller **255** directly. In like manner, a microcontroller **257** is used in the slave interface **250***d* to communicate information to and from the internal bus **205**, from and to a Bluetooth base station **258**, for wireless communication with Bluetooth-compliant devices via an RF network **170'**. Techniques for transforming data to and from an internal bus **205** and an external network **150'**, **170'**, **201**, are common in the art.

[0022] In a preferred embodiment of this invention, a processor **220** receives information from a UPnP user control point (UCP) **120** via an IP network interface device **210** and the internal bus **205**. The interface device **210** includes Ethernet, xDSL, cable modem, wireless local loop, satellite, fiber to curb, or other IP network structure. The processor **220** transforms the UPnP information from the UCP into network and device specific commands, and communicates these network and device specific commands, as required, via the internal bus **205**, to the appropriate slave interface device **250***a-d* for communication to the particular non-UPnP device being controlled. In like manner, the processor receives information from each non-UPnP device, or from a network controller of the non-IP network, via the slave interface device **250***a-d*, transforms the information into UPnP messages, as required, and communicates these UPnP messages to the UCP **120**. Other embodiments will be evident to one of ordinary skill in the art. For example, in a USB-specific embodiment, the processor **220** may communicate directly with the IP network interface **210** for communicating UPnP messages, and directly with a USB host controller **255** for communicating USB messages, without the need for an intermediate bus structure **205**.

[0023] The specific functions performed by the processor **220** to support UPnP messaging are discussed further below with regard to **FIG. 5**. **FIGS. 3 and 4** are presented to provide a context for the understanding of the functions presented in **FIG. 5**.

[0024] The UPnP Device Architecture defines the protocols for communication between UPnP user control points (UCPs) and devices. **FIG. 3** illustrates the UPnP protocol stack **300** that is used for the discovery, description, control,

eventing, and presentation phases of UPnP network management. At the highest layer **310**, messages contain only UPnP vendor-specific information about their devices. Moving down the stack, vendor content **310** is supplemented by information **320** defined by UPnP Forum working committees. Messages from the layers **310, 320** above are hosted in UPnP-specific protocols **330**, defined by the UPnP architecture. These protocols **330** are formatted using the Simple Service Discovery Protocol (SSDP), General Event Notification Architecture (GENA), and Simple Object Access Protocol (SOAP), and delivered via HTTP, at level **340**. The HTTP **340** is either multicast **342** or unicast **344** running over UDP **352**, or standard HTTP **346, 348** running over TCP **354**. Each UDP **352** or TCP **354** message, at protocol level **350**, is delivered via IP **360**.

[0025] **FIG. 4** illustrates an example UPnP process for establishing and maintaining a network of UPnP user control points and controlled devices. The foundation for UPnP networking is IP addressing. Each device is assigned a unique address, at **410**, either via an assignment by a Dynamic Host Configuration Protocol (DHCP) server that is managing the network, or via an Auto IP address generation function, if the network is not managed. Devices may also be assigned a device name, for ease of subsequent references to each device.

[0026] Given an IP address, the next step in the UPnP process is discovery **420**, wherein each device provides the network with a few essential specifics about the device or its services, with a pointer to more detailed information, as required. The UCPs may also use the discovery process to search for devices of particular interest. The devices advertise their essential characteristics when they first enter the network, as well as in response to a search for their characteristics by a UCP. To assure that the network is kept up to date, devices are required to periodically refresh their advertisement via the discovery process **420**. Devices are logged off the network when they communicate a logoff message, or when they fail to refresh their advertisement.

[0027] The next step in the UPnP process is description **430**, wherein UCPs that are interested in advertised devices issue a request for additional information from a URL (Universal Resource Locator) address that is contained in the device advertisement. Typically, this additional information regarding the device and its services is located at the device, but it may also be located at a remote location, such as an Internet site that is maintained by the vendor of the device.

[0028] When a UPnP UCP learns of a device's capabilities, it is able to control and/or monitor the device, at **440**, via an action request or a value query. In response to the action request, the device effects the action, and reports a result. Generally, the result is an acknowledgement that the requested action was effected, but it may be a more detailed message that reports the current device state, and/or the state of one or more variables associated with the device. In response to a value query, the device reports the state of one or more variables identified in the value query.

[0029] The UCP may also request notification whenever an event occurs at the device, via the eventing process **450**. The UCP 'subscribes' to be notified of any change of state at the device, and may exclude specified state changes, such as the change of value of particular variables, from this notification process. Whenever a device changes state, it notifies all subscribers of the event, except those subscribers that have excluded the specific state change from their subscription.

[0030] The UCP presents the capabilities and controls associated with a device, based on a presentation page that is provided by the device, at **460**. The UCP requests the presentation page from a URL that is provided in the device description. As with the device description at **430**, the URL may address the device, or it may address a remote site, such as the vendor's Internet site, or a third-party service provider's site.

[0031] **FIG. 5** illustrates an example block diagram of a UPnP enabling device **200** comprising a UPnP interface **210**, a UPnP proxy enabling processor **220** (**220***a* and **220***b*), and an interface **250** to a non-IP network in accordance with this invention.

[0032] The UPnP (IP network) interface **210** includes an IP network module **501**, and a network services layer **502** for accessing the IP network module **501**, including creating and managing network communications, formatting appropriate IP messages, and receiving and sending messages. Consistent with conventional practice, the network services layer **502** sends multicast UDP messages multiple times, to enhance reliability.

[0033] The UPnP HTTP server **503** is a server process that supports the HyperText Transfer Protocol (HTTP) used for communication between the UPnP user control points (UCPs) **120** and the controlled devices (**150-180** in **FIG. 1**), as discussed above with regard to the HTTP protocol layer **340** of **FIG. 3**. In a preferred embodiment, the HTTP server **503** handles interactions between multiple UCPs **120** and multiple devices, and is configured to provide a non-blocking transfer. This non-blocking transfer is easily effected via the use of threads to handle different types of requests, as discussed further below. The functions provided by a HTTP server **503** in a preferred embodiment include:

[0034] creating and managing threads to handle device connect and disconnect, and to handle UPnP defined queries for device capability, description, and presentation;

[0035] creating and maintaining a network table **504** that keeps track of each network and the type of threads created for the network, and records the communication data structures for each thread;

[0036] monitoring a pre-defined TCP/IP server port and a pre-defined multicast UDP port to receive HTTP messages and to pass them to the corresponding modules that are responsible for the messages; and

[0037] providing the Application Program Interface (API) for transforming responses and GENA notifications into proper HTTP messages, and invokes network services **502** to send the messages.

[0038] The UPnP HTTP server **503** uses the network table **504** and the value of the HTTP request line, such as the HTTP requests GET, POST, M-POST, M-SEARCH, SUBSCRIBE, and UNSUBSCRIBE for dispatching. For example, upon receipt of an HTTP M-SEARCH request, it dispatches messages to the discover server modules **510**

corresponding to each network in the UPnP enabling device **200**, to effect the requested search.

[0039] In a preferred embodiment of the UPnP enabling device **200**, the processor **220** includes two parts for interfacing with the UPnP interface **210**. A first part **220***a* includes components that are embodied for each slave network or each device, and a second part **220***b* includes components that are embodied for each service provided by each slave device in each slave network. For example, a VCR device typically provides a variety of services, including a clock service, a tuner service, and a tape transport service.

[0040] The network-level processing block **220***a* includes the modules **510**, **520**, **530** required to effect and coordinate the UPnP discovery, presentation, and description phases, respectively, as well as a device manager module **540** that effects and coordinates commands and messages related to each device in the slave network. A device connect/disconnect handler **550** provides information to the appropriate databases **515**, **525**, **535** that the modules **510**, **520**, **530** use to respond to UPnP requests regarding the presence of devices on the network, and their capabilities. In a preferred embodiment, the device connect/disconnect handler **550** provides the following functions:

[0041] determining the devices connected to the associated slave network;

[0042] loading the code and data required for each connected device; and

[0043] providing device-dependent data and code to the modules **510**-**530**, via the databases **515**-**535**. In general, the device-dependent data and code is provided via access to a file server **130**, discussed further below.

[0044] When activated, the device connect/disconnect handler **550** uses the slave network interface **250** to determine the identity of each device in its associated network. In accordance with one aspect of this invention, a file server **130** is accessible via the IP network interface **210**. This file server **130** is configured to contain the detailed information required to effect the UPnP notification, coordination, and control functions for each identified device, as well as the mapping between the advertised UPnP commands and the corresponding device and network specific commands. Depending upon the available memory (**230** in **FIG. 2**) at the UPnP enabling device **200**, the processor **220** fills in the discovery, presentation, and description information at the databases **515**, **525**, **535**, respectively. Alternatively, the processor **220** merely stores the appropriate URLs of each device's presentation and description information, for subsequent communication to the UCP **120**, as required, and as discussed above. These URLs may address information on the file server **130**, or at other accessible locations, such as a vendor-supported, or third-party provided, Internet site.

[0045] The device connect/disconnect handler **550** accesses this information from the file server **130** via a device code and data loader **590** that is configured to form the appropriate IP-compatible requests, and to receive the corresponding IP responses. The particular functions that the device code and data loader **590** provide depend upon the distribution of information storage between the UPnP enabling device **200** and the file server **130**, and includes some or all of the following functions:

[0046] constructing the local path of the URL associated with each device's code and data, based on the IP address or host name of the file server **130**;

[0047] providing the interface to the device connect/disconnect handler **550** to facilitate sending notifications regarding the need to access the code or data associated with a particular device;

[0048] forming the HTTP/GET message to fetch the required code or data for the device, via the UPnP HTTP server **503**;

[0049] receiving the results of the HTTP/GET message from the UPnP HTTP server **503**; and

[0050] returning the results to the device connect/disconnect handler **550**.

[0051] If the UPnP enabling device **200** allows for dynamically loaded code to support the slave device or network interface, the code and data loader **590** also separates the code and data, communicates the data to the device connect/disconnect handler **550**, processes the code, and stores it to a memory address assigned for dynamic linking. In a preferred embodiment, to conserve memory space (**230** in **FIG. 2**) at the device **200**, the file server **130** contains the binary code for each process that is potentially required at the device **200**. This code is preferably stored and communicated as an attachment to an HTML page that is associated with the particular device and/or function.

[0052] In a preferred embodiment, after creating and starting one device connect/disconnect handler **550** for each slave network, the HTTP server **231** is placed in a wait state during initialization until at least one of the handlers have finished adding the required information to the corresponding databases. After initialization, the handler **550** monitors each device for connection and disconnection, and updates each database **515**, **525**, **535** by appropriately adding or deleting device information. The handler **550** also forms one or more GENA notification messages, and invokes the API of the HTTP server **503** to multicast such additions and deletions. The handler **550** also periodically forms an SSDP 'alive' message, and invokes the API of the HTTP server **503** to broadcast the message, thereby refreshing each device's active status on the IP network.

[0053] The discovery server module **510**, and corresponding device capability database **515**, implement the UPnP discovery server specification. As noted above, in a preferred embodiment, the discovery module **510** is responsible for providing the UPnP discovery function for each device within its corresponding network. The functions of the discover module **510** in a preferred embodiment include:

[0054] providing an API for querying the network or devices for device characteristics;

[0055] processing UPnP search messages, such as an M-SEARCH message with an "ssdp:discover" message header; and

[0056] upon receipt of an SSDP query, searching the device capability database **515**, forming a response, and invoking the aforementioned HTTP server **503** API to return the response to the requester.

[0057] The device capability database **515** contains data structures in memory that store information about the capabilities of each device known to the module **510**, and is preferably organized for efficient operations for SSDP searches.

[0058] The description server module **530** implements the UPnP description server specification, discussed above. The module **530** either provides the appropriate URL for locating the device description and/or the presentation, or it provides the device description and/or the presentation, directly or via the file server **130**, for devices that do not have a corresponding remote URL address at which the description and/or the presentation is located. Initially, it will be expected that devices on a non-IP network will not have an associated UPnP description at a remote URL address, and thus the UPnP enabling device **200** will need to provide the description, via a device description database **535**, or via access to the file server **130**. As this invention becomes commonplace, however, vendors or third party developers are likely to develop UPnP descriptions for non-UPnP devices, and the amount of information required to be stored at the device description database **535**, or at the file server **130** will, correspondingly, be substantially reduced. The functions of the description server module **530** include:

[0059] providing an API for querying device descriptions;

[0060] processes HTTP/GET messages addressed to the local description server that is responsible for presenting the description for the devices on the slave network under its responsibility; and

[0061] searching the device description database **535** in response to HTTP/GET messages, and invoking the API at the HTTP server **503** to return the response.

[0062] The presentation module **520** implements the UPnP presentation server specification, and is configured similar to the description server module **530** to respond to HTTP/GET messages addressed to the local presentation server responsible for the devices on the network, using the device presentation database **525**, or the file server **130** as required.

[0063] In a preferred embodiment, the device manager module **540** enables multiple UCPs **120** to simultaneously control multiple devices in the slave network under its responsibility, in response to device access and control requests, such as HTTP POST and M-POST messages. The functions of the device manager module include:

[0064] creating and managing threads to route and handle device control requests, as discussed below; and

[0065] providing an interface for the device connect/disconnect handler to provide notification of device connect and disconnect events.

[0066] The device table **545** stores the mapping between a service identification (for example, a device UUID and a service name) and the data structures used to communicate data with the service control server **570** and the event subscription server **560**.

[0067] The service-level UPnP block **220***b* includes an event subscription server module **560**, a service control server module **570**, and an event source module **580**. Typically, a device provides one or more services. Preferably, there is one event subscription server module **560**, one service control server module **570**, and one event source module **580** associated with each service provided by a device. Correspondingly, there is one event subscription database **565** and one service state table **585** associated with each service.

[0068] The service control server module **570** is responsible for effecting control commands directed to its associated service. The functions of the service control server module **570** in a preferred embodiment includes:

[0069] parsing SOAP commands, invoking the appropriate driver interface(s) to effect each command, and invoking the API at the HTTP server **503** to send an acknowledgement or failure message to the requester;

[0070] updating the service state table **585** upon successful command execution, if the state of the service changes;

[0071] monitoring events posted by the slave device, and updating the service state table **585** if the state of the service changes; and

[0072] invoking the event source module **580** with each update of the service state table **585**.

[0073] In a preferred embodiment, because not all slave device drivers are configured to report the entire state of the driven device, the service state table **585** is used to record the current value of the state of each service (power, register values, and so on). The table **585** is initialized when the device enters the UPnP control network and is kept consistent with the state of the service(s) provided by the device by updating the state every time a state-changing command is successfully executed.

[0074] The event subscription server module **560** is responsible for allowing UCPs to express their interest about device events related to each service. The functions of the event subscription server module **570** in a preferred embodiment includes:

[0075] parsing GENA event subscription messages, entering the subscribing UCP's identification and subscribed events in the event subscription database **565**, or at the file server **130**, and invoking the API of the HTTP server **503** to send an acknowledgement (or failure notification) to the subscriber UPnP controller; and

[0076] invoking the event source module **580** to pass the current state of the service to a first-time subscriber UCP.

[0077] The event source module **580** is responsible for posting events of the service to all UCPs that have subscribed to such events. The functions of the event source module **580** in a preferred embodiment includes:

[0078] providing an interface for the service control module **570** to pass notifications about the changes in the service status to the service state table **585**;

[0079] examining the event subscription database **565**, or the corresponding data on the file server **130**,

notifying subscriber UCPs of subscribed event changes by forming a GENA notification message, and invoking the API of the HTTP server **503** to send the GENA message; and

[0080] providing an interface for the event subscription server module **560** to effect the notification of each first-time subscriber of the state of the service, via the formation and transmission of a GENA notification message, via the API of the HTTP server **503**.

[0081] **FIG. 6** illustrates an example flow diagram of thread creation to provide a non-blocking architecture for communications between the UCPs and the slave devices, in accordance with this invention. For convenience and ease of understanding, the foregoing description provides references to items in the previous figures, although the principles presented in this flow diagram are also applicable to other structures or system configurations. The first digit of each reference numeral corresponds to the first figure at which the referenced item is introduced.

[0082] At **610**, the HTTP server **503** allocates and initializes memory spaces for the network table **504**, the device capability database **515**, the device description database **535**, and the device presentation database **525**, for each slave network. As noted above, this initialization information may include references to information that is stored at the file server **130**, or at remote URLs. The HTTP server **503** also allocates and initializes a space for communication and synchronization between itself and each of the slave network's device connect/disconnect handler **550**. At **615**, the HTTP server **503** creates a device connect/disconnect handler thread for each network, and waits until at least one of the device connect/disconnect handlers **550** reports that it has successfully initialized the device capability database **515**, the device description database **535**, and the device presentation database **525**. When the HTTP server **503** receives the notification that the device connect/disconnect handler **550** has initialized the databases **515**, **525**, **535**, the HTTP server **503** allocates and initializes a data structure for each working thread that it will create, at **620**. These data structures are used to communicate with the threads. The HTTP server **503** repeats the process **615-620** for each network, as each network's device connect/disconnect handler **550** reports a successful initialization of the network's databases **515**, **525**, **535**. At **630**, the HTTP server **503** creates working threads, one for handling device discovery, one for handling device description, and one for handling device presentation. Each thread activates the corresponding modules and receives a pointer to the database **515**, **535**, and **525**, respectively, that it will use. At **635**, the HTTP server **503** records each network type, each thread type, and the communication data structure for each thread, into the network table **504**. Thereafter, the HTTP server **503** directs each device manager **540** to set up service handling threads for each device in the network for which the manager **540** is responsible. The manager **540** executes in the context of the HTTP server **231**.

[0083] At **650**, each device manager **540** first queries the discovery service module **510** to obtain a list of devices in the network for which it is responsible. For each device, the manager further queries the description server module to get a list of services provided by the device. The manager **540** then creates a service-handling thread for each service provided by each device, and a corresponding data structure for communicating with each thread. At **655**, the device manager **540** records the mapping of each thread to each service provided by the device in the device table **545**.

[0084] At **670**, each service-handler thread allocates and initializes the event subscription database **565** and the service state table **585** for its associated service. At **675**, each service-handler thread activates each of the service control **570**, event subscription **560**, and event source **580** modules associated with the service.

[0085] Not illustrated, when a device is added to the network, the device manager **540** creates and records a service-handler thread for each service provided by the device, as in blocks **650-655**. The newly created service-handler thread creates and initializes the service-specific database **565** and table **585**, and activates the modules **560**, **570**, **580**, as in blocks **670-675**, above.

[0086] At **690**, all threads created in blocks **630** and **650** wait until being notified of pending work, via the data structure associated with each thread. When the HTTP server **503** identifies an incoming request for a particular working thread, the server **503** places the request into the data structure corresponding to the thread, then returns to handle the next request. In this manner, the HTTP server **503** devotes substantially little time to the processing of request; the actual processing of each request is effected via a single placement of the request into an appropriate data structure. In a preferred embodiment, each thread periodically checks the contents of its data structure. When one or more items of the data structure change, the thread determines the appropriate action to take in response to the change, and reacts accordingly. After the work is completed, the thread invokes the API at the HTTP server **503** to communicate an acknowledgement (or a failure notice if the request was not fulfilled) to the UCP that issued the incoming request. In the case of an incoming control command, the command is placed in communication data structure of the service-handling thread of the targeted service. When the service-handling thread detects this command in its data structure, it determines the type of command. If the command is an event subscription, it passes the command to the event description server module **560**. If the command is a service control command, the command is passed to the service control server module **570**.

[0087] Alternative thread initiation and control schemes will be readily apparent to one of ordinary skill in the art. For example, a thread can be created when a request for a particular service arrives for the first time. In this scheme, for example, the device manager **540** provides an interface for the device description server module **530** to pass a notification when a description is requested by a UCP. Upon receiving the notification, the device manager **540** checks the device table **545** to determine if the service-handling thread already exists for the device; if not, a thread is created for each service provided by the device. In this manner, service-handling threads are only created for devices for which at least one UCP has expressed interest. Alternatively, although threads may be expected to provide an efficient implementation, processes can be used to implement the enabling logic in lieu of threads. Such processes will communicate either via shared memory, as in the case of threads, or via message passing.

[0088] As presented above, an embodiment of this invention provides a means for facilitating the control of non-UPnP devices via a UCP. As will be evident to one of ordinary art, if, as in the examples provided, shared memory is used for communication and synchronization, proper locking mechanisms, common in the art, should be used to ensure proper operation. It is important, for example, for the device capability database **515**, the device description database **535**, the device presentation database **525**, and the device table **545** to be consistent, and therefore atomic operations for updating each database should be enforced. For example, write operations to a database or table will typically take priority over read operations, to assure that the read operation is provided the freshest data. These and other means of maintaining data consistency are common in the art.

[0089] In a preferred embodiment of this invention, the use of a consistent naming convention scheme is used to simplify the design. For example, the local part of the URL that is used for each server has the prefix: network_type/server_type, such as "usb/descriptionServer", or "bluetooth/presentationServer", and so on. To facilitate locating of device files at the file server **130** by the device connect/disconnect handler **550**, each file name contains an identifier of the device, and the contents of the file, such as "USB interface.code", "laser_printer.description", or "scanner.capability". These names may be made more specific by including, for example, an indication of the make or model of the device. If device functions are provided via library functions, the function names contain a prefix that uniquely identifies the device, thereby avoiding function names conflicts.

[0090] The foregoing merely illustrates the principles of the invention. It will thus be appreciated that those skilled in the art will be able to devise various arrangements which, although not explicitly described or shown herein, embody the principles of the invention and are thus within its spirit and scope. For example, the particular functional partitioning presented in the figures is presented for illustrative purposes, and various combinations of hardware and software implementations may be used to embody the invention. These and other system configuration and optimization features will be evident to one of ordinary skill in the art in view of this disclosure, and are included within the scope of the following claims.

I claim:

1. A UPnP interface device that is configured to facilitate UPnP control of at least one non-UPnP device that are located on one or more slave networks using one or more different network technologies, comprising:

an IP interface to at least one UPnP controller, the UPnP controller being configured to issue a UPnP command in conformance with a UPnP protocol,

at least one non-IP interface to the at least one non-UPnP device, and

a processor that is configured to:

receive the UPnP command, via the IP interface,

transform the UPnP command into a device command,

communicate the device command to a target device of the at least one non-UPnP device via the at least one non-IP interface, and

communicate a UPnP acknowledgement of the UPnP command to the at least one UPnP controller, via the IP interface, and

2. The device of claim 1, wherein the one or more network technologies include at least one of: a USB network, a bluetooth network, a HAVi-compatible network, an IEEE 1394 network, a Home API network, a HomeRF network, a Firefly network, a power line network, an X-10 network, and a Jini-compatible network.

3. The device of claim 1, wherein:

the UPnP controller is further configured to issue a UPnP request in conformance with the UPnP protocol,

the UPnP request includes one of: a description request, a presentation request, a subscription request, and a query, and

the processor is configured to provide at least one of: a device description, a service description, a presentation page, an event, and a value of a variable, in response to the UPnP request.

4. The device of claim 3, wherein

the IP interface also provides access to a file server, and

the processor provides the at least one of: the device description, a service description, a presentation page, an event, and a value of a variable, based on information received from the file server.

5. The device of claim 1, wherein

the processor includes at least one of:

a discovery module that is configured to provide an advertisement of the at least one non-UPnP device to the UPnP controller,

a description module that is configured to provide a description of functions of the at least one non-UPnP device to the UPnP controller, in response to a request from the UPnP controller, and

a presentation module that is configured to provide a presentation page that facilitates a control of the at least one non-UPnP device by a user.

6. The device of claim 5, wherein

at least one of the discovery module, the description module, and the presentation module is configured to provide the advertisement, the description, and the presentation page, respectively, for the at least one non-UPnP device of the slave networks.

7. The device of claim 1, wherein

the processor includes at least one of:

a service control module that communicates commands to the target device,

an event subscription module that receives requests from the at least one UPnP controller to be notified of one or more changes of state of the target device, and

an event source module that notifies the at least one UPnP controller of one or more changes of state of the target device.

8. The device of claim 7, wherein

the service control module maintains a service state table that reflects the state of the target device, and

the event source module notifies the at least one UPnP controller of the one or more changes of the state of the target device based on the service state table.

9. The device of claim 1, wherein the UPnP server enabler communicates the device command to the target command by modifying a data structure that is associated with a thread, and the thread effects the communication to the at least one non-UPnP device of the slave networks.

10. The device of claim 1, wherein

the IP interface also provides access to a file server, and

the processor effects the transform of the UPnP command into the device commands based on information received from the file server.

11. The device of claim 1, wherein

the processor is further configured to recognize a connection and disconnection of the at least one non-UPnP device, and initiates and terminates threads in response to each connection and disconnection, respectively.

12. A method for facilitating UPnP control of at least one non-UPnP device on a slave network, comprising:

receiving device-dependent data corresponding to each of the at least one non-UPnP device from a file server,

receiving a UPnP command in conformance with a UPnP protocol from a UPnP controller,

transforming the UPnP command into a device command, based on the device-dependent data received from the file server,

communicating the device command to a target device of the at least one non-UPnP device on the slave network, and

communicating a UPnP acknowledgement of the UPnP command to the UPnP controller.

13. The method of claim 12, wherein the slave network is one of: a USB network, a bluetooth network, a HAVi-compatible network, an IEEE 1394 network, a Home API network, a HomeRF network, a Firefly network, a power line network, an X-10 network, and a Jini-compatible network.

14. The method of claim 12, further including:

receiving a UPnP request in conformance with the UPnP protocol,

the UPnP request including one of: a description request, a presentation request, a subscription request, and a query, and

providing at least one of: a device description, a service description, a presentation page, an event, and a value of a variable, in response to the UPnP request, based on information received from the file server.

15. The method of claim 12, further including at least one of:

providing an advertisement of at least one non-UPnP device to the UPnP controller,

providing a description of functions of the at least one non-UPnP device to the UPnP controller, in response to a request from the UPnP controller, and

providing a presentation page that facilitates a control of the at least one non-UPnP device by a user.

16. The method of claim 15, wherein

at least one of the advertisement, the description, and the presentation page are provided by the file server.

17. The method of claim 12, further including

receiving requests from the UPnP controller to be notified of one or more changes of state of the at least one non-UPnP device, and

notifying the UPnP controller of one or more changes of state of the at least one non-UPnP device.

18. The method of claim 17, further including

maintaining a service state table that reflects the state of the target device, and

notifying the UPnP controller of the one or more changes of the state of the at least one non-UPnP device based on the service state table.

19. The method of claim 12, further including

creating a thread that is associated with the at least one non-UPnP device of the slave network, and

modifying a data structure that is associated with the thread; and

wherein the thread is configured to effect the communication of the device command to the at least one non-UPnP device of the slave network, based on the modification of the data structure.

20. A network comprising:

an IP sub-network,

a non-IP sub-network, and

a UPnP enabling device that facilitates control of a device on the non-IP sub-network by a UPnP controller on the IP sub-network.

21. The network of claim 20, further including

a file server on the IP sub-network, and

wherein

the UPnP enabling device facilitates the control of the device based on information received from the file server.

22. The network of claim 20, wherein

the UPnP enabling device is configured to:

receive a UPnP command from the UPnP controller on the IP network,

transform the UPnP command into a device command, and

communicating the device command to the device on the non-IP sub-network.

23. The network of claim 22, wherein

the UPnP enabling device is further configured to provide at least one of: a device description, a service description, a presentation page, an event, and a value of a

variable corresponding to the device on the non-IP network, in response to a UPnP request from the UPnP controller.

24. The network of claim 23, further including

a file server on the IP sub-network, and

wherein

the UPnP enabling device provides the at least one of: the device description, the service description, the presentation page, the event, and the value of the variable, based on information received from the file server.

25. The network of claim 20, wherein

the UPnP enabling device facilitates the control of the device on the non-IP sub-network by a UPnP controller on the IP sub-network via the use of threads that provide a non-blocking communication.

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US005825894A

United States Patent [19]

Shennib

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[45] **Date of Patent: Oct. 20, 1998**

[54] **SPATIALIZATION FOR HEARING EVALUATION**

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[52] **U.S. Cl.** **381/60**; 381/17; 600/559

[58] **Field of Search** 381/17, 68.2, 68.4, 381/60, 68.1; 128/746; 600/559

[56] **References Cited**

U.S. PATENT DOCUMENTS

2,394,569	2/1946	Strommen .	
4,118,599	10/1978	Iwahara et al.	179/1
4,139,728	2/1979	Haramoto et al.	179/1
4,219,696	8/1980	Kogure et al.	179/1
4,759,070	7/1988	Voroba et al.	381/60
4,774,515	9/1988	Gehring	242/53
4,809,708	3/1989	Geisler et al.	128/746
4,901,353	2/1990	Widin .	
5,173,944	12/1992	Begault	381/17
5,233,665	8/1993	Vaughan et al.	381/17
5,303,306	4/1994	Brillhart et al. .	
5,325,436	6/1994	Soli et al. .	
5,434,924	7/1995	Jampolsky	387/17
5,436,975	7/1995	Lowe et al.	381/17

FOREIGN PATENT DOCUMENTS

US96/13126 6/1996 WIPO .

OTHER PUBLICATIONS

Mueller, H. Gustav; "A Practical Guide to Today's Bonanza of Underused High-Tech Hearing Products," The Hearing Journal, Mar. 1993, vol. 46, No. 3, pp. 13–27.

Hawkins, David B., Ph.D.; Probe Microphone Measurements: Hearing Aid Selection and Assessment, Chapter 5, "Prescriptive Approaches to Selection of Gain and Frequency Response", pp. 91–112.

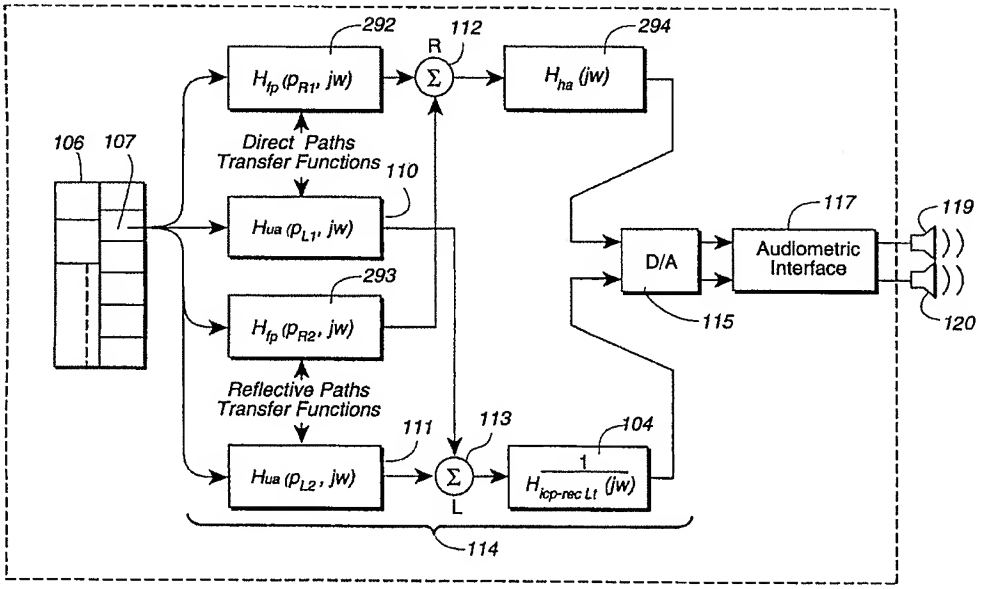
(List continued on next page.)

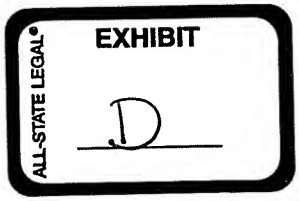
Primary Examiner—Forester W. Isen
Attorney, Agent, or Firm—Michael A. Glenn

[57] **ABSTRACT**

A hearing evaluation and hearing aid fitting system provides a fully immersive three-dimensional acoustic environment to evaluate unaided, simulated aided, and aided hearing function of an individual. Digital filtering of one or more signal sources representing speech and other audiologically significant stimuli according to selected models and digitally controlled signal processing parameters, including audio sources, spatializing coordinates, acoustic boundaries, signals representing one or more simulated hearing aids, and individualized body/external ear transfer functions synthesizes a simulated acoustic condition for presentation to a hearing-impaired person for objective and subjective hearing evaluation via an intra-canal prosthesis that is positioned in the ear canal, and that incorporates a microphone probe to measure in-the-ear-canal responses at a common reference point near the tympanic membrane during unaided, simulated aided, and aided hearing evaluation, thus providing measurements that are directly correlated across all phases of hearing assessment during the fitting process of a hearing aid.

4 Claims, 28 Drawing Sheets





OTHER PUBLICATIONS

Mowrer, Donald E., Ph.D. and Steams, Carol; "Threshold measurement variability among hearing aid dispensers," Hearing Instruments, vol. 43, No. 4, 1992, pp. 26–27.

Gauthier, E.A. and Rapisardi, D.A., MS; "A threshold is a threshold is a threshold . . . or is it?", Hearing Instruments, vol. 43, No. 3,1992, pp. 26–27.

Earphone Calibration, "Real Ear Methods", p. 26.

Valente, M; Potts, L.; and Vass, Bill; "Intersubject Variability of the Real–Ear SPL: TDH–39P vs ER 3A Earphones."

Chen, Joseph K. and Geisler, C. Daniel, "Estimation of eardrum acoustic pressure and of ear canal length from remote points in the canal", Acoust. Soc. Am. 87 (3), Mar. 1990, pp. 1237–1247.

Sandberg, Robert, MS; McSpaden, Jay B., Ph.D.; and Allen, Dan, MA; "Real measurement from real ear equipment," Hearing Instruments, vol. 42, No. 3, 1993, pp. 17–18.

"Selection instrumentation/master hearing aids in review," Hearing Instruments, vol. 39, No. 3, 1988, p. 18–20.

"Its—Hearing Aid Simulator," software brochure.

Wightman, Frederic L. and Kistler, Doris J.; "Headphone simulation of free–field listening. I: Stimulus synthesis,". Acoust. Soc. Am. 85(2), Feb. 1989, pp. 858–867.

Wightman, Frederic L. and Kistler, Doris J.; "Headphone simulation of free–field listening. II: Psychophysical validation,". Acoust. Soc. Am. 85(2). Feb. 1989, pp. 868–878.

Cherry, E. Colin; "Som Experiments on the Recognition of Speech, with One and with Two Ears," The Journal of the Acoustical Society of America, vol. 25, No. 5, Sep. 1953, pp. 975–979.

Cherry, E. Colin and Taylor, W.K.; "Some Further Experiments upon the Recognition of Speech, with One and with Two Ears," The Journal of the Acoustical Society of America, vol. 26, No. 4, Jul. 1994, pp. 554–559.

Bronkhorst, A.W. and Plomp, R.; "The effect of head–induced interaural time and level differences on speech intelligibility in noise," J. Acoust. Soc. Am. 83 (4), Apr. 1988, pp. 1508–1516.

Bronkhorst, A.W. and Plomp, R.; "Effect of multiple speech-like maskers on binaural speech recognition in normal and impaired hearing,". Acoust. Soc. Am. 92 (6), Dec. 1992, pp. 3132–3139.

Begault, Durand R. and Wenzel, Elizabeth M., NASA Ames Research Center, Moffett Field, California, "Headphone Localization of Speech," Human Factors, 1993, 35(2), 361–376.

American National Standard, Specification of Hearing Aid Characteristics, ANSI S3.22–1987.

"three–dimensional audio for PC–compatibles," The Beachtron, Beachtron User's Guide, Crystal River Engineering, Inc.

Zdeblick, Mark J. Ph.D., "A Revolutionary Actuator for Microstructures," reprint from Sensors, Feb. 1993.

Mills, A.W., "On the Minimum Audible Angle,". Acoust. Soc. Am., vol. 30, No. 4, Apr. 1953, pp. 237–246.

Rife, Douglas D. and Vanderkooy, John; "Transfer–Function Measurement with Maximum–Length Sequences," J. Audio Eng. Soc., vol. 37, No. 6, Jun. 1989, pp. 419–442.

American National Standard, Specification for Audiometers, ANSI S3.6. 1989.

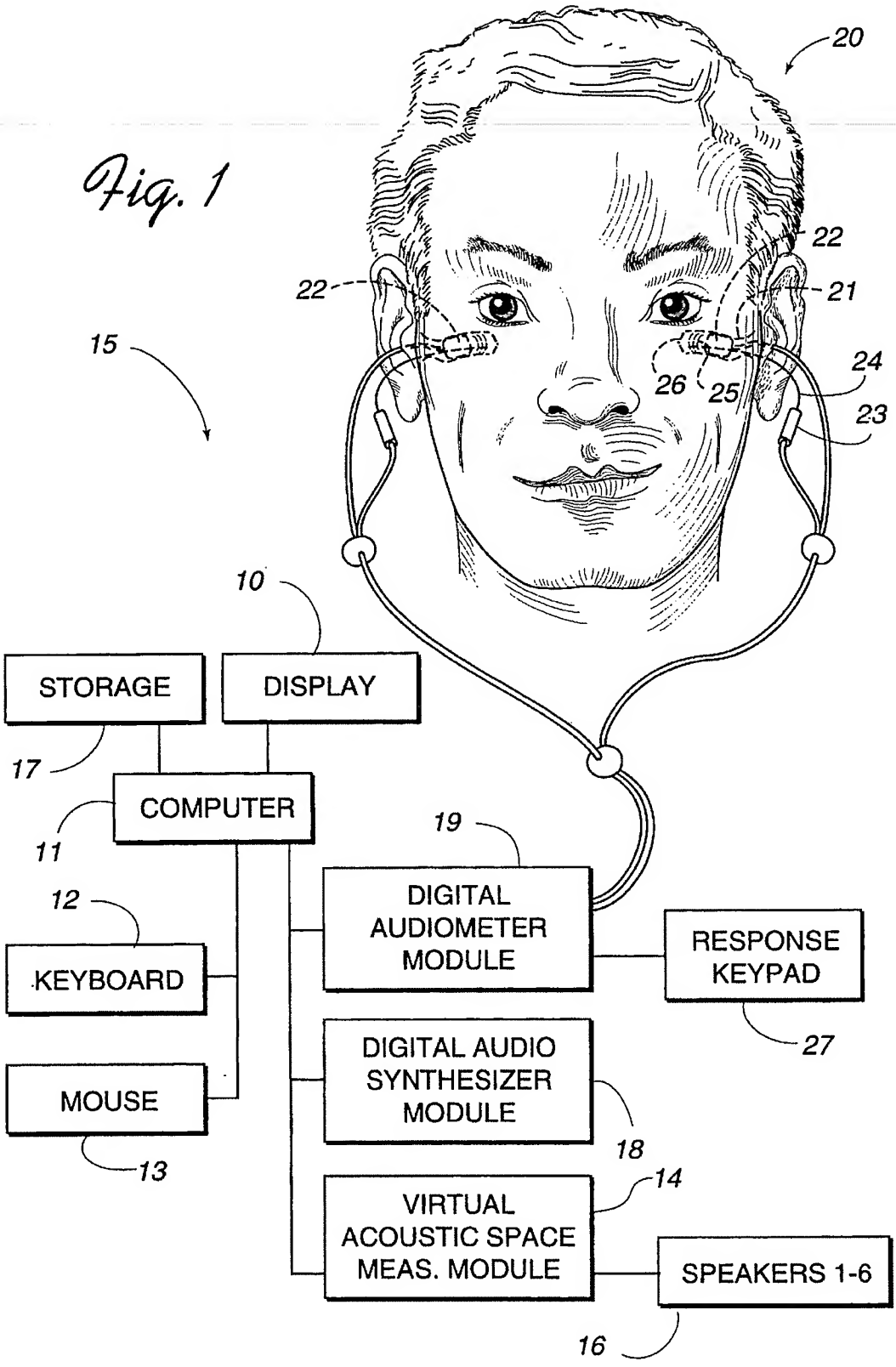
Jamieson, Donald G., "Consumer–Based Electroacoustic Hearing Aid Measures," JSLPA Monogr. Suppl. 1, Jan. 1993, pp. 87–98.

Assessment of Fitting, Arrangements, Special Circuitry, and Features, pp. 221–224.

Wenzel, "Localization in Virtual Acoustic Displays", Presence, vol. 1, No. 1, pp. 81–107, 1992.

Begault et al, "Call Sign Intelligibility Improvement Using A Spatial Auditory Display," (NASA Technical Memorandum 104014) Apr. 1993.

Fig. 1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STORAGE
DISPLAY
17
COMPUTER
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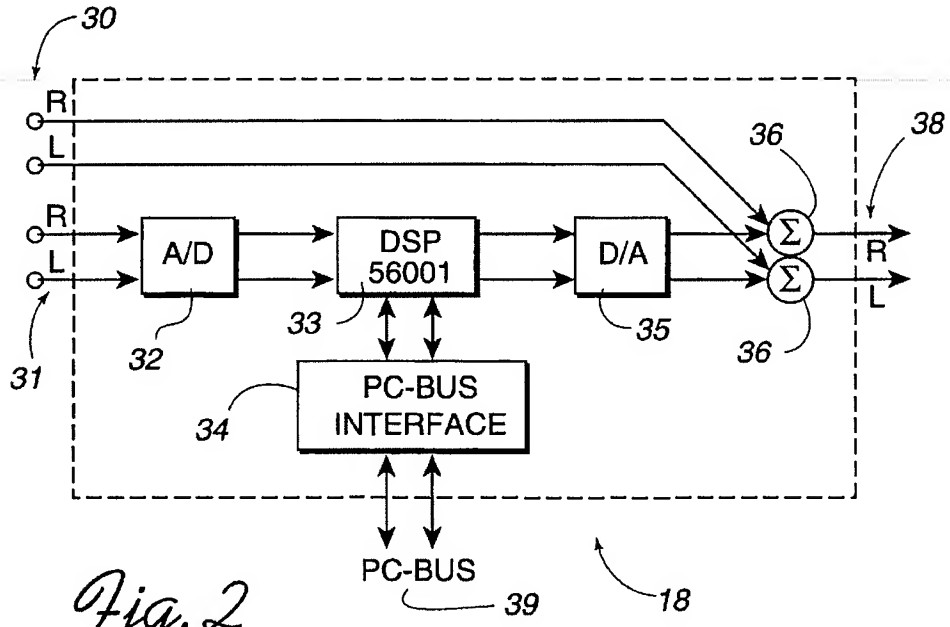
12
DIGITAL AUDIOMETER MODULE
RESPONSE KEYPAD
KEYBOARD
27
DIGITAL AUDIO SYNTHESIZER MODULE
MOUSE
18
13
14
VIRTUAL ACOUSTIC SPACE MEAS. MODULE
SPEAKERS 1-6
16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D/A
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PC-BUS
INTERFACE
34
PC-BUS
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Fig. 2

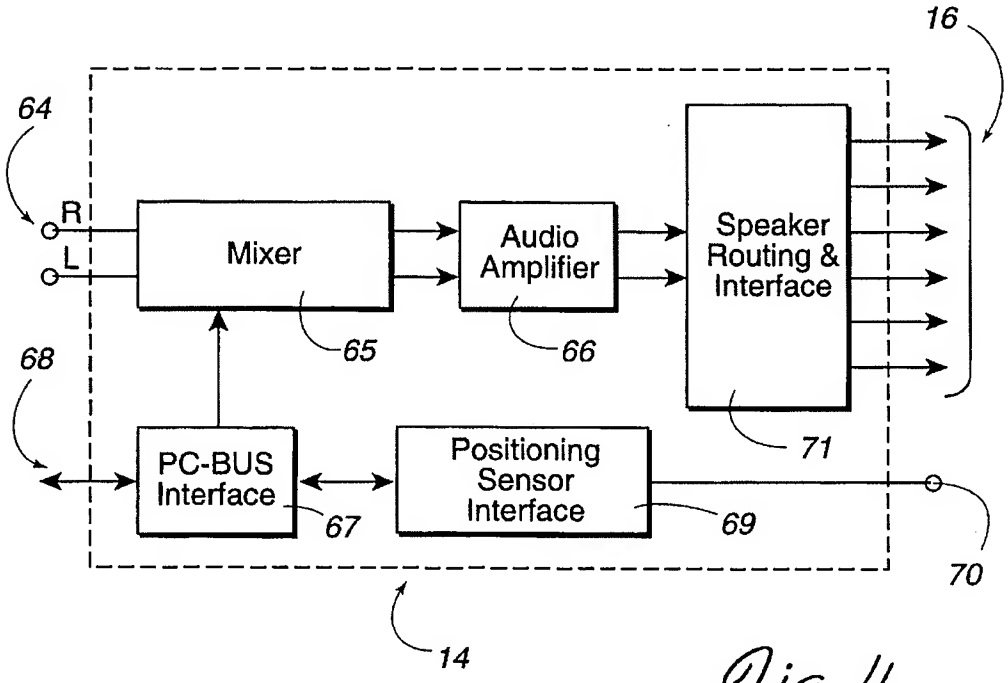
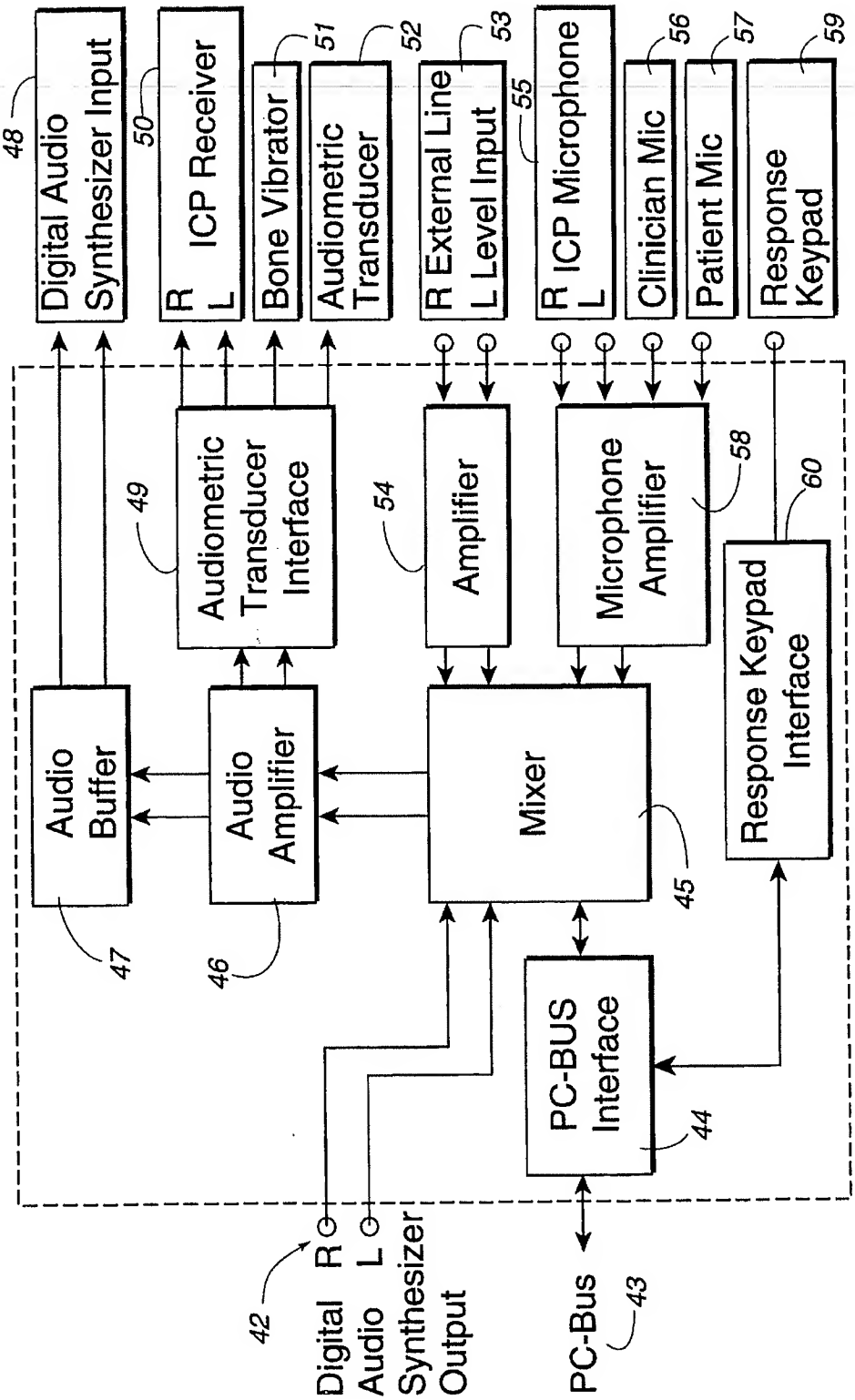

16
64
R
L
Mixer
Audio
Amplifier
Speaker
Routing &
Interface
65
66
71
68
PC-BUS
Interface
67
Positioning
Sensor
Interface
69
70
14

Fig. 4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48
R ICP Receiver L
50
Bone Vibrator
51
Audiometric Transducer
52
R External Line L Level Input
53
55
R ICP Microphone L
Clinician Mic
56
Patient Mic
57
Response Keypad
59
19
Audiometric Transducer Interface
49
Amplifier
54
Microphone Amplifier
58
60
Response Keypad Interface
Audio Buffer
47
Audio Amplifier
46
Mixer
45
PC-BUS Interface
44
Digital Audio Synthesizer Output R L
42
PC-Bus
43

Fig. 3

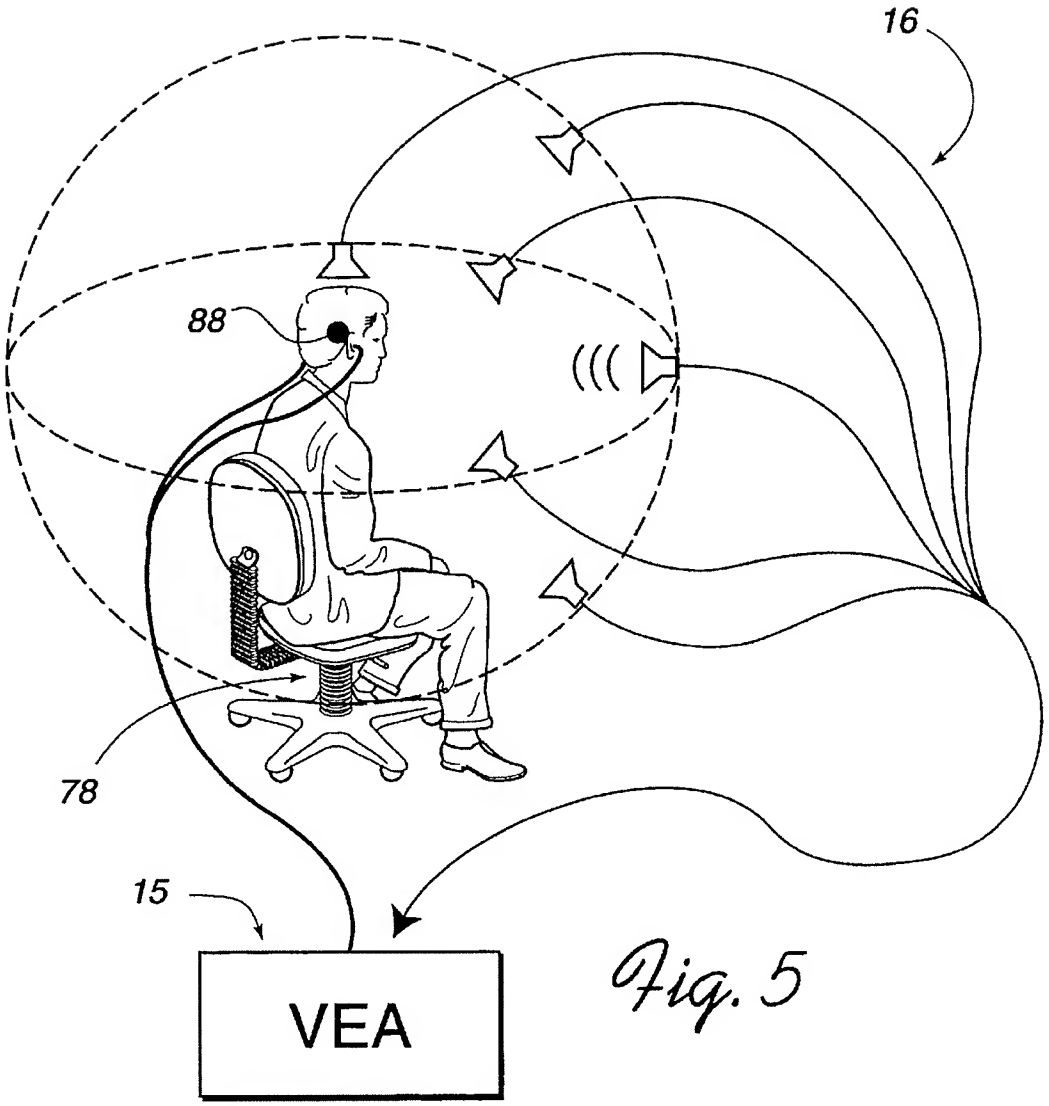

Fig. 5

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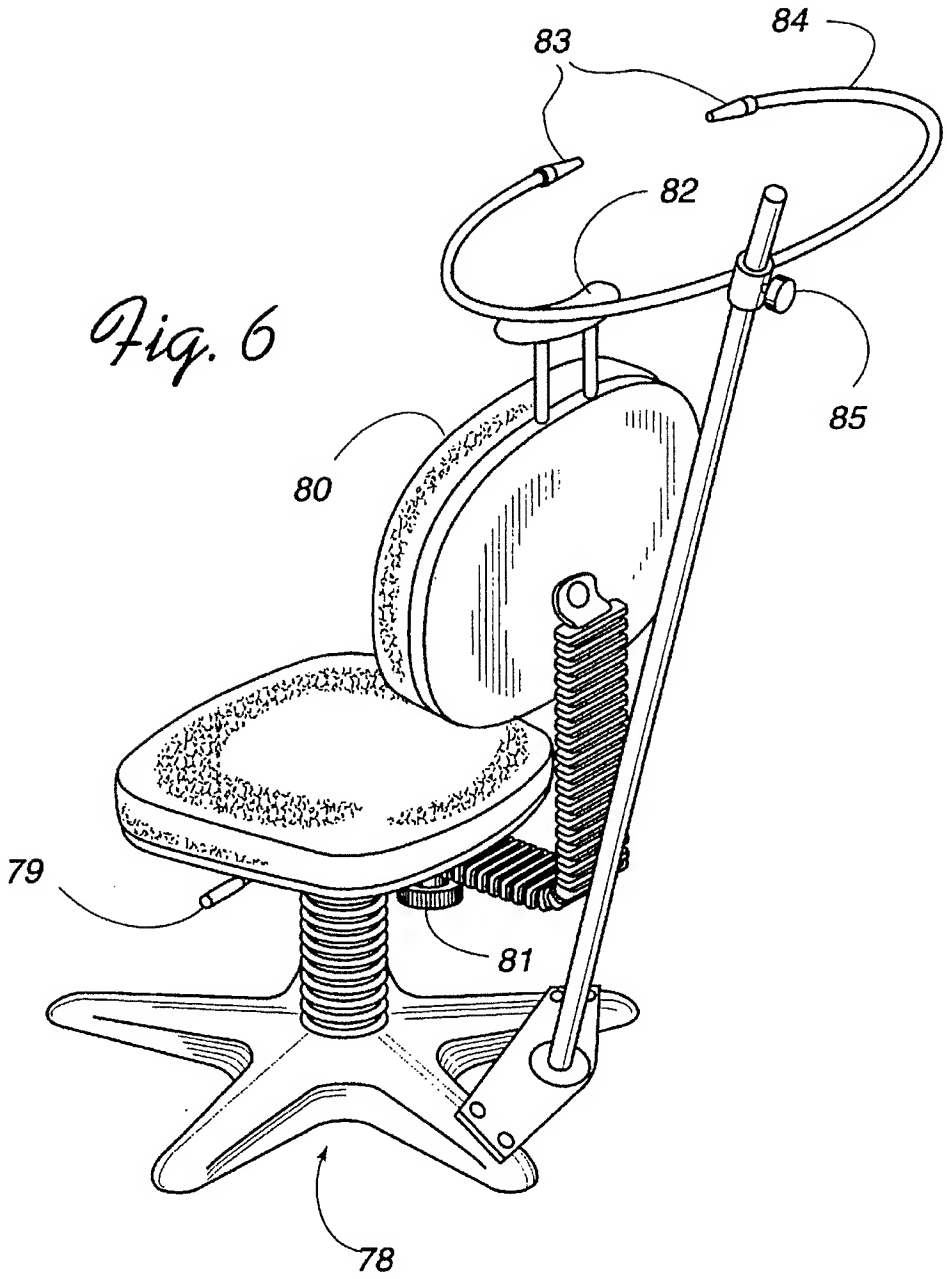

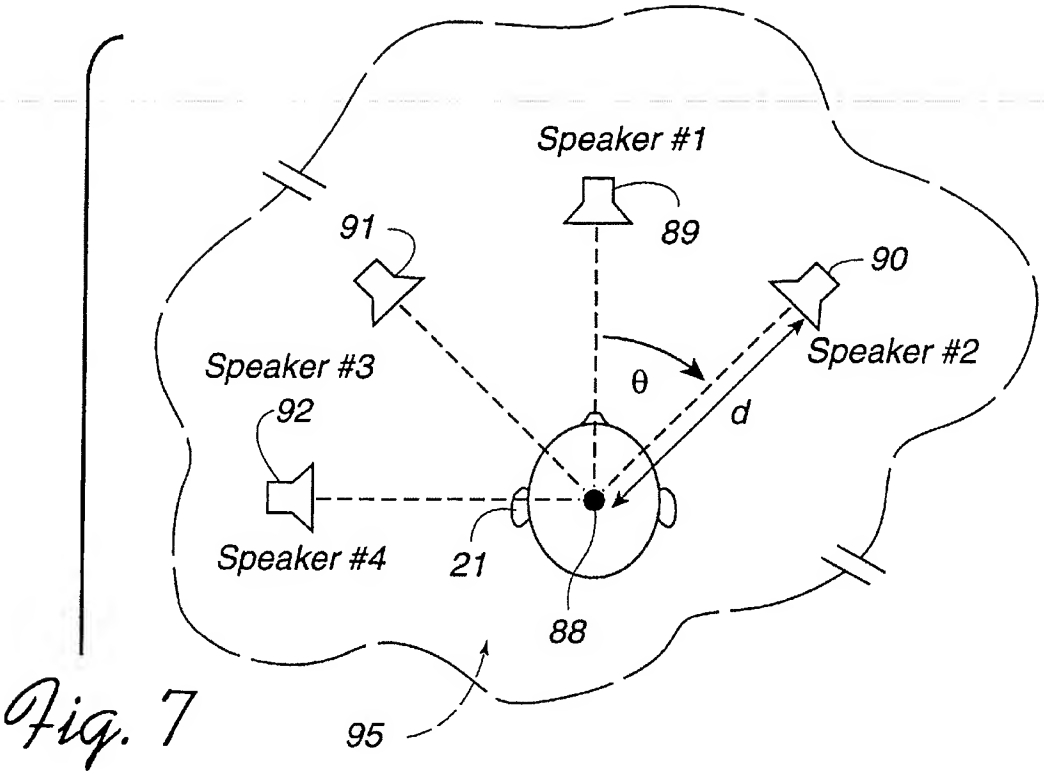
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Speaker #3
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Speaker #2
d
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Speaker #4
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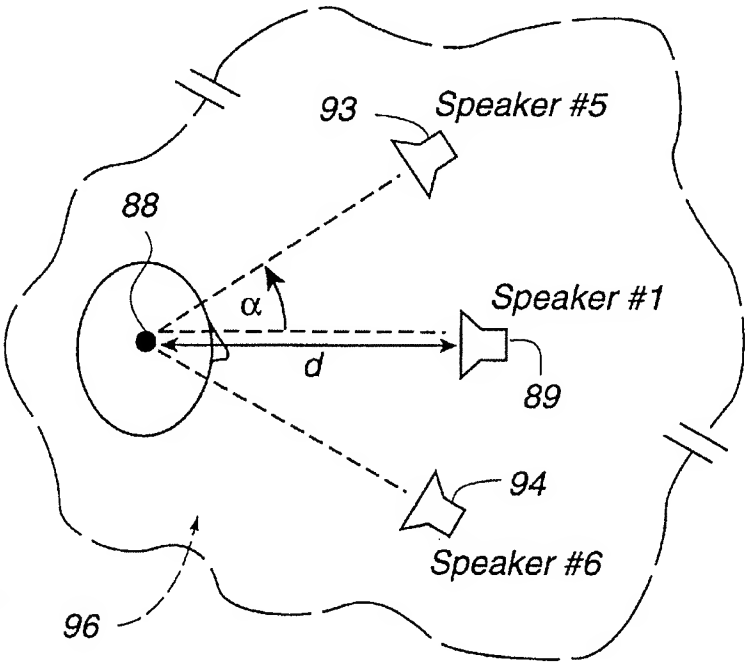
Fig. 7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α
Speaker #1
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Speaker #6
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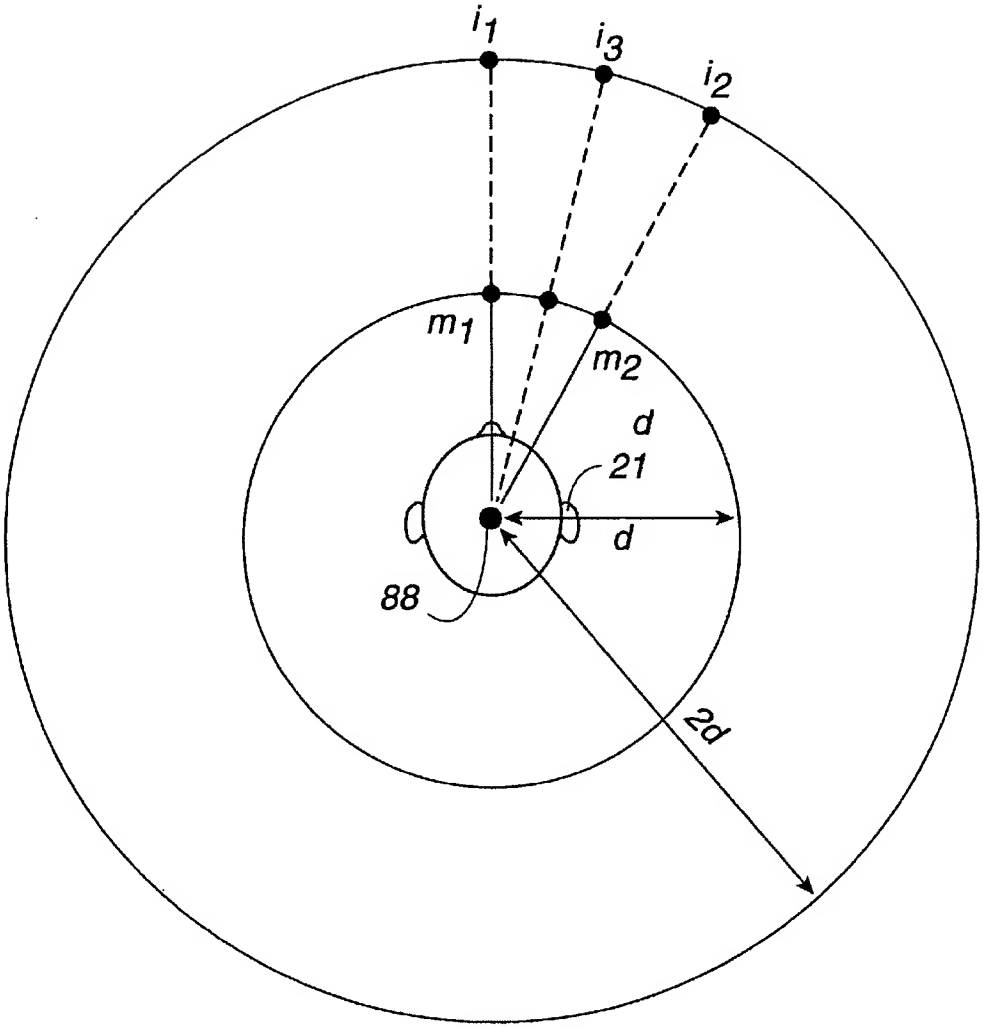

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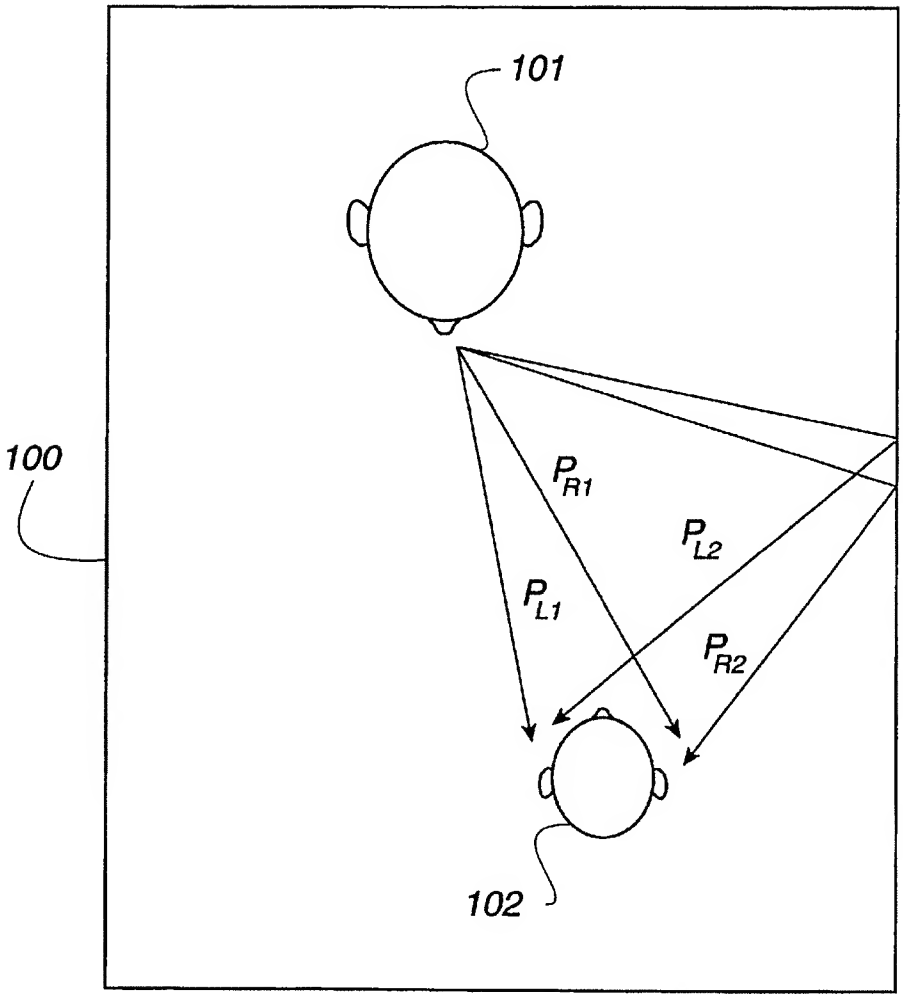

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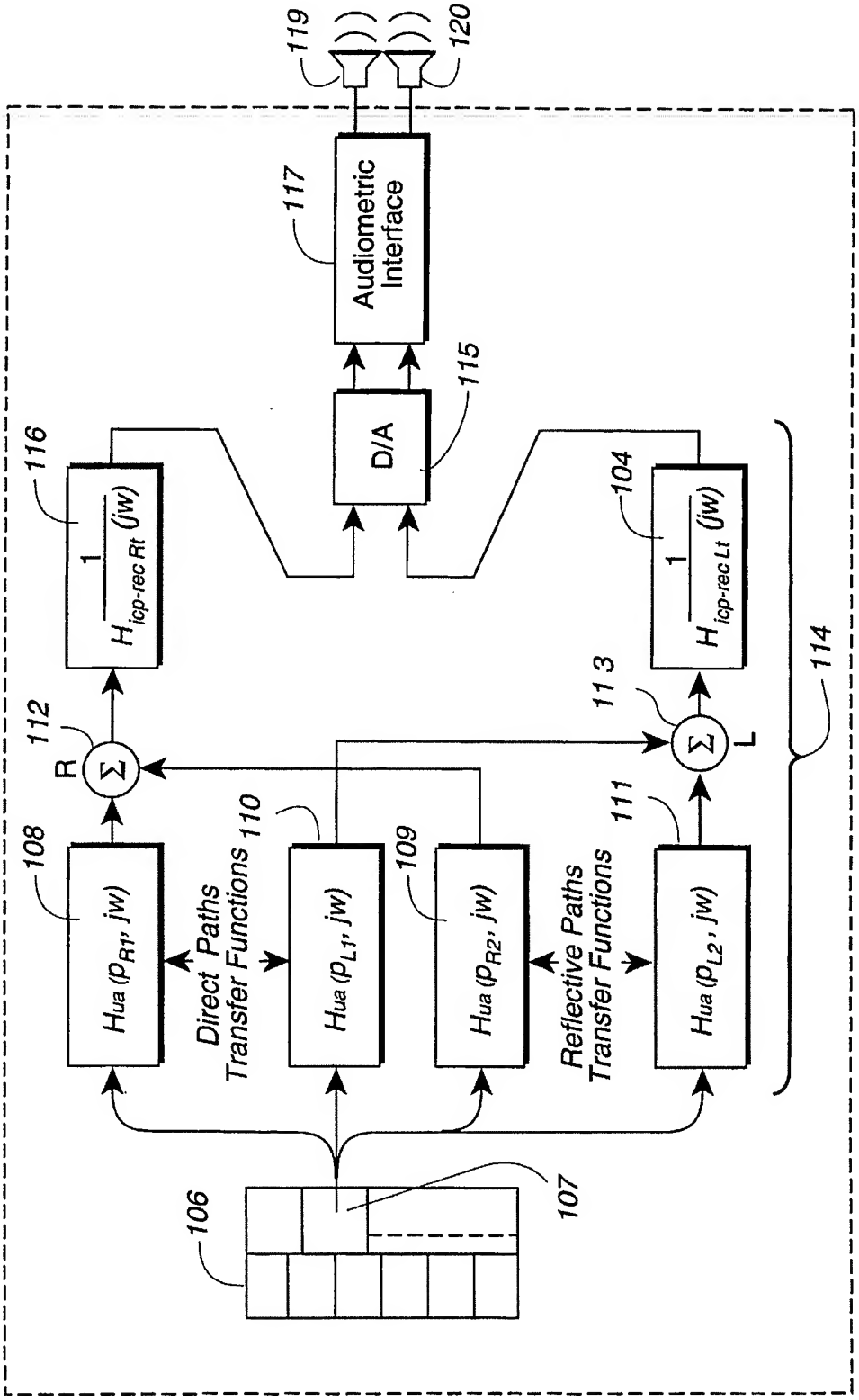
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Audiometric Interface
115
D/A
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$\frac{1}{H_{icp\text{-}rec\ Rt}(jw)}$
104
$\frac{1}{H_{icp\text{-}rec\ Lt}(jw)}$
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Σ
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Σ
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$H_{ua}(\rho_{R1}, jw)$
Direct Paths Transfer Functions
110
$H_{ua}(\rho_{L1}, jw)$
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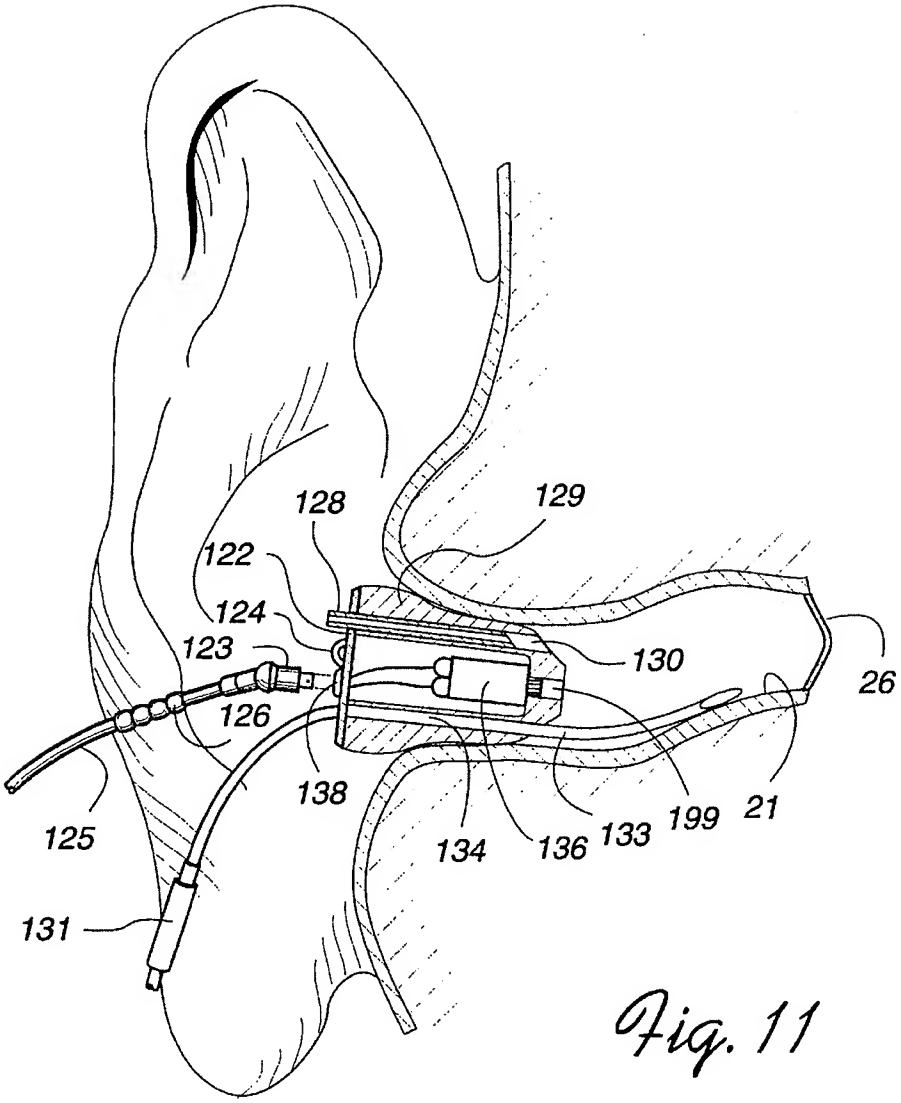
$H_{ua}(\rho_{R2}, jw)$
Reflective Paths Transfer Functions
111
$H_{ua}(\rho_{L2}, jw)$
106
107

Fig. 10



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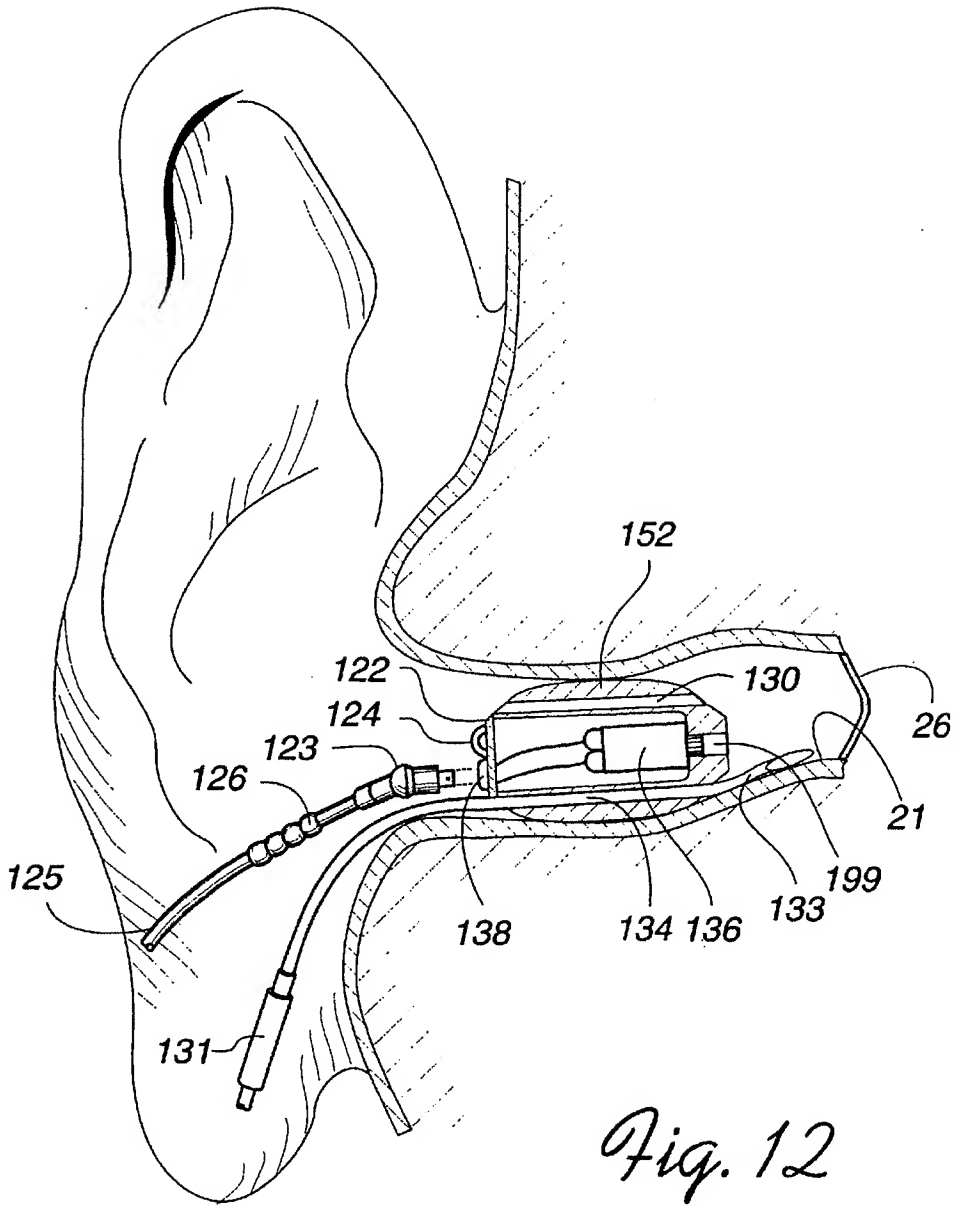

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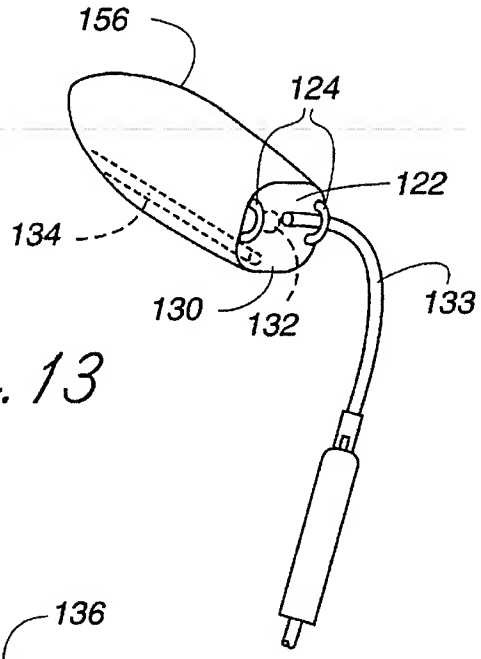

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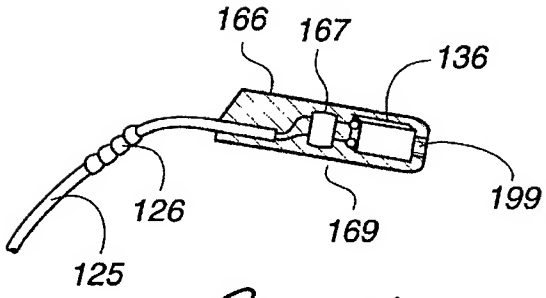

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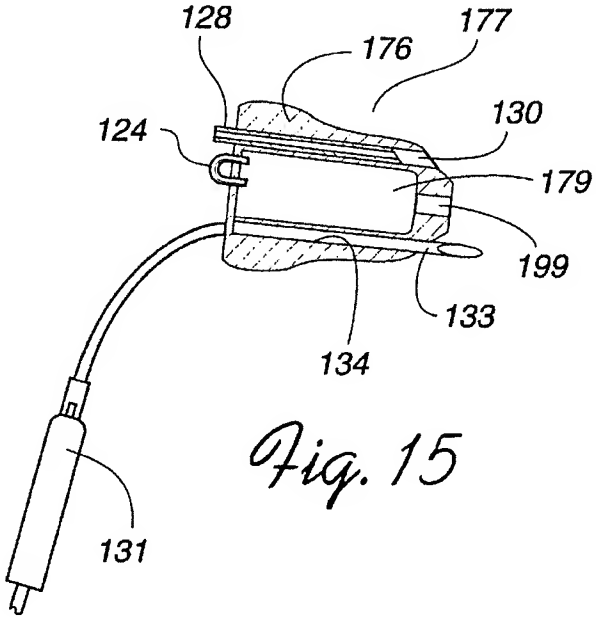

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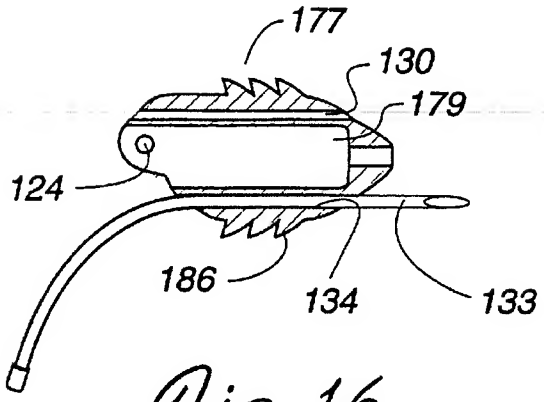

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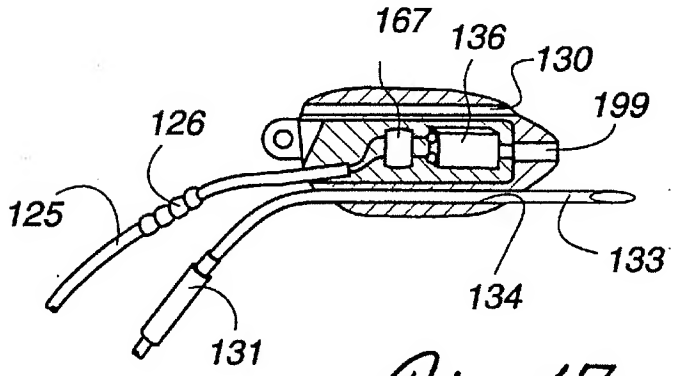

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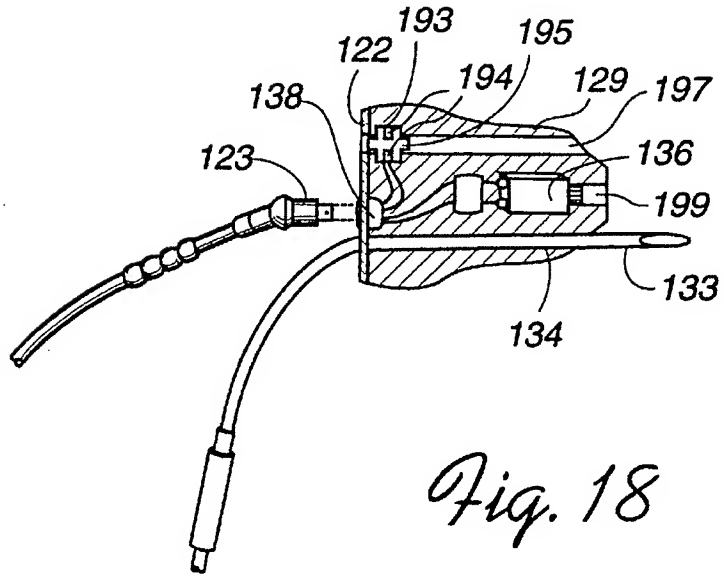

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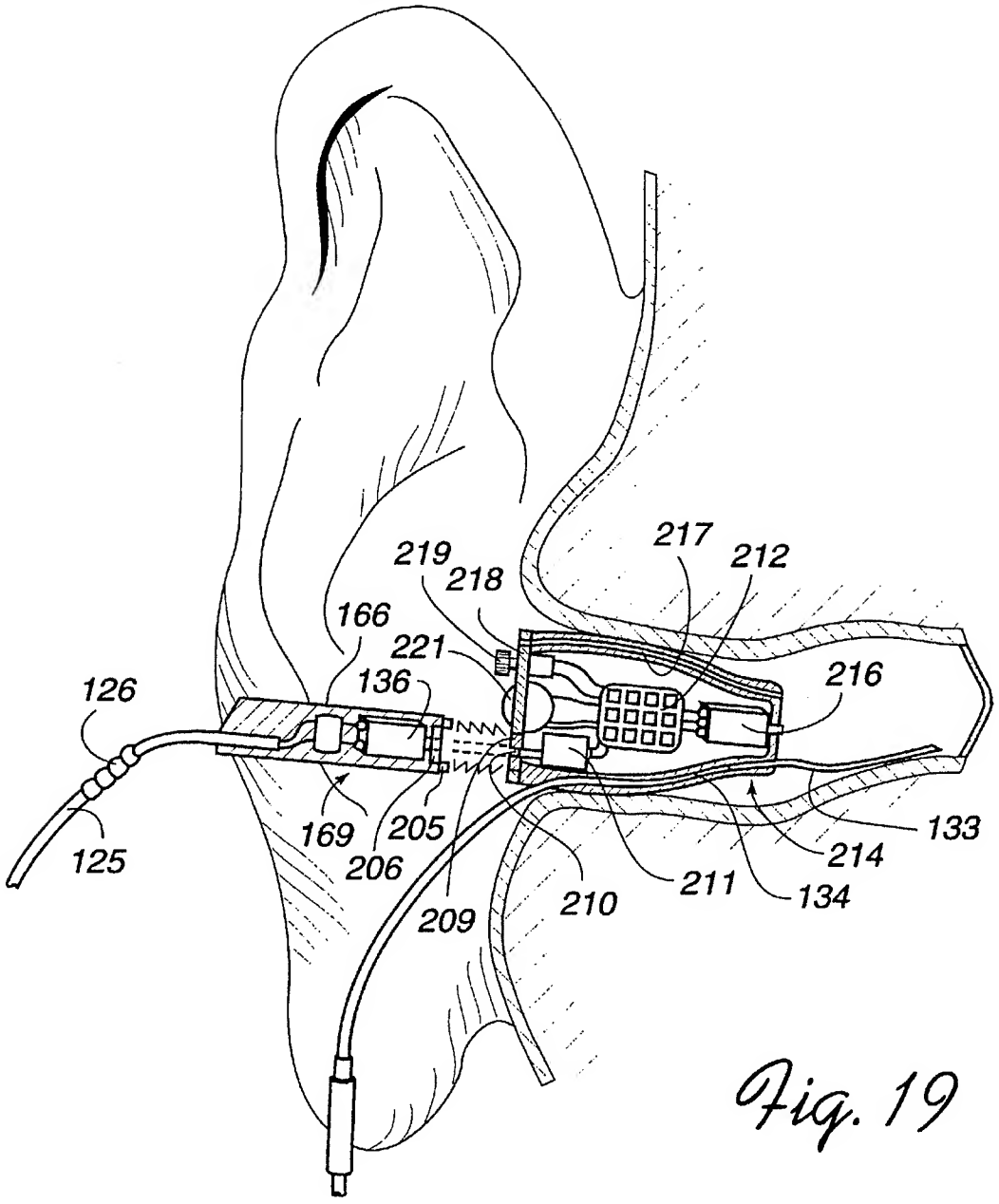
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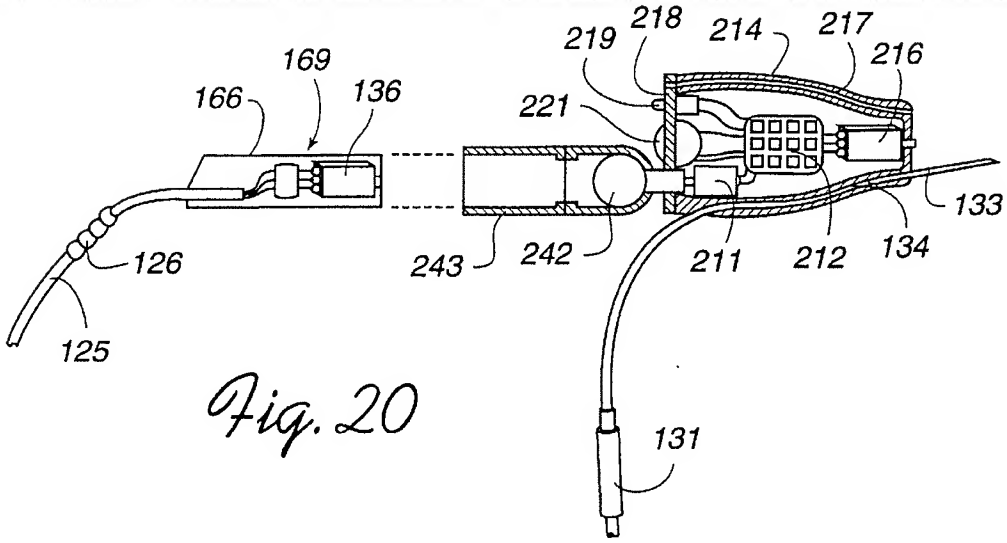
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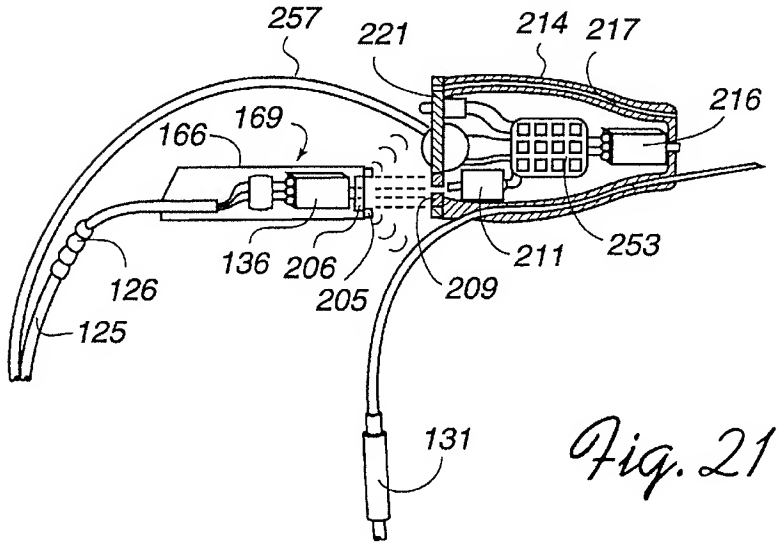
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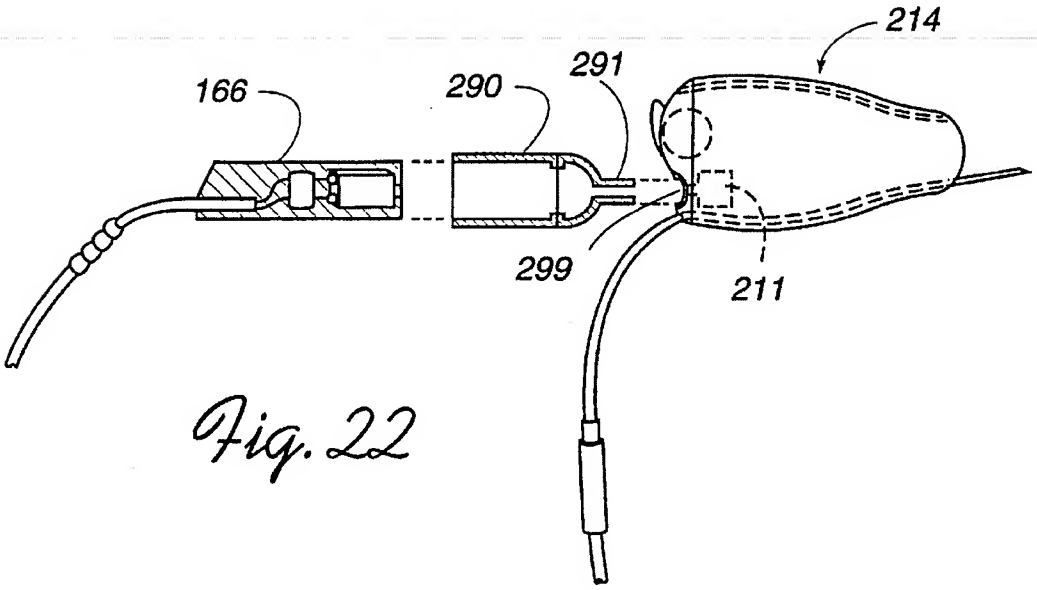
Fig. 21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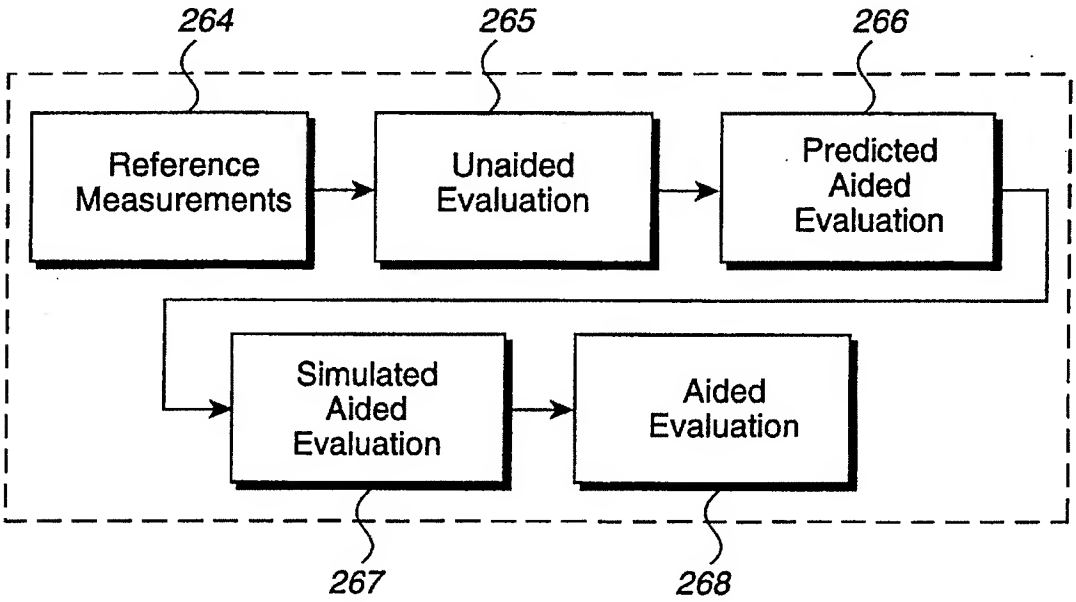
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Reference Measurements
Unaided Evaluation
Predicted Aided Evaluation
Simulated Aided Evaluation
Aided Evaluation
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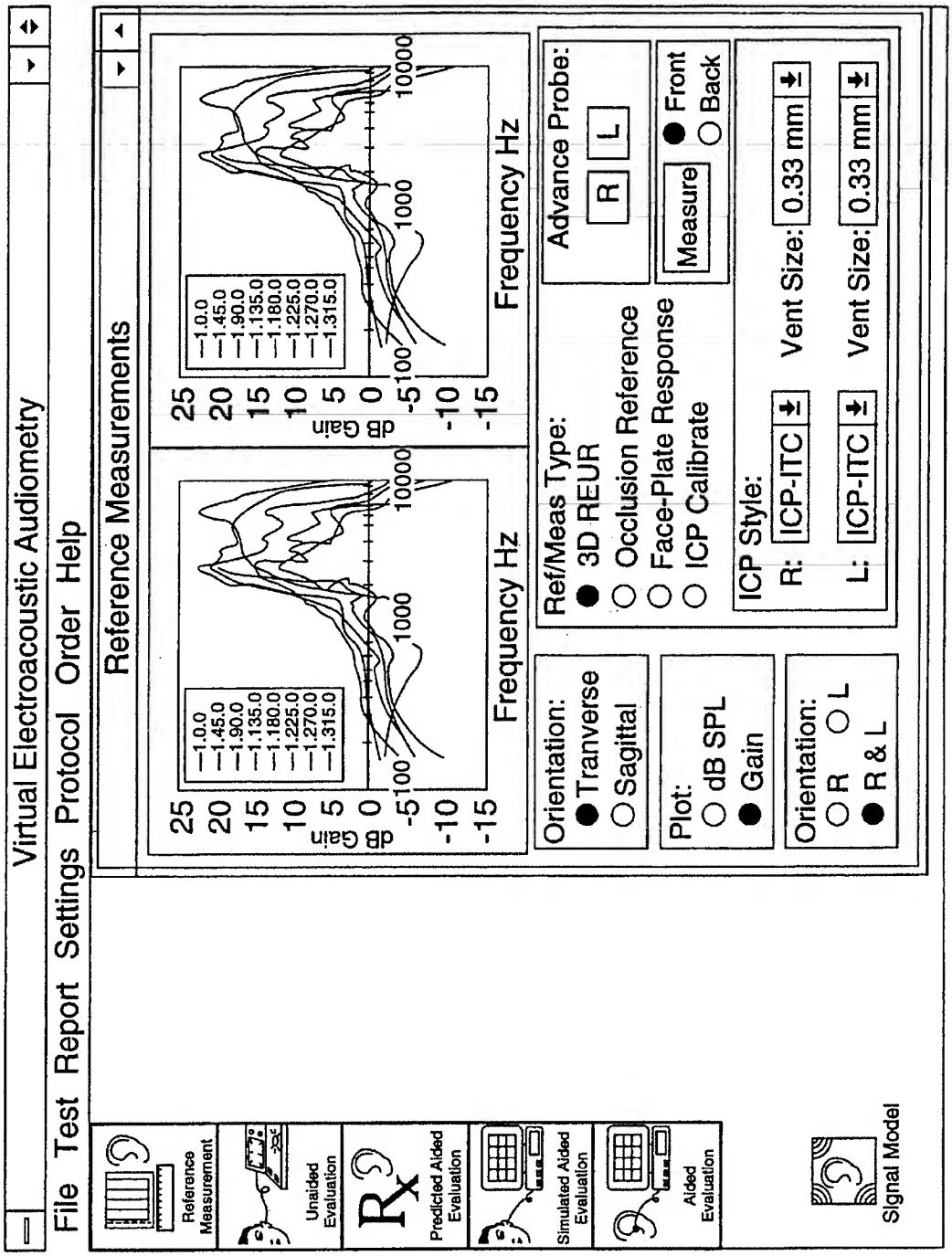
Fig. 23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Reference Measurement
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Simulated Aided Evaluation
Aided Evaluation
Signal Model
dB Gain
Frequency Hz
1.0.0
1.45.0
1.90.0
1.135.0
1.180.0
1.225.0
1.270.0
1.315.0
Orientation:
Tranverse
Sagittal
Plot:
dB SPL
Gain
Orientation:
R L
R & L
Ref/Meas Type:
3D REUR
Occlusion Reference
Face-Plate Response
ICP Calibrate
Advance Probe:
R L
Measure
Front
Back
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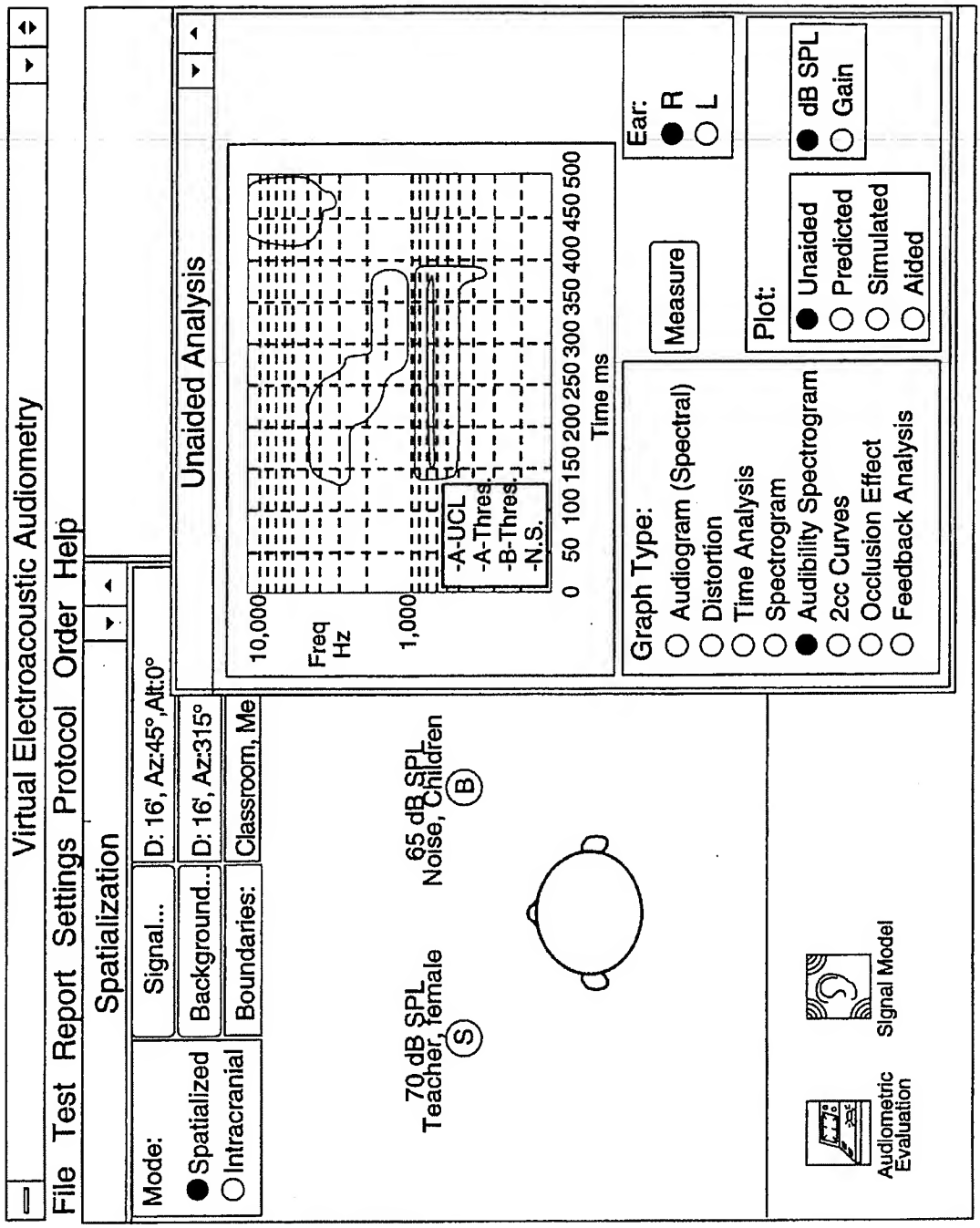
Vent Size: 0.33 mm
L: ICP-ITC
Vent Size: 0.33 mm

Fig. 24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Spatialization
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Spatialized
Intracranial
Signal...
Background...
Boundaries:
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D: 16', Az:315°
Classroom, Me
Unaided Analysis
10,000
Freq Hz
1,000
-A-UCL
-A-Thres.
-B-Thres.
-N.S.
0 50 100 150 200 250 300 350 400 450 500
Time ms
Graph Type:
Audiogram (Spectral)
Distortion
Time Analysis
Spectrogram
Audibility Spectrogram
2cc Curves
Occlusion Effect
Feedback Analysis
Measure
Ear:
R
L
Plot:
Unaided
Predicted
Simulated
Aided
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Gain
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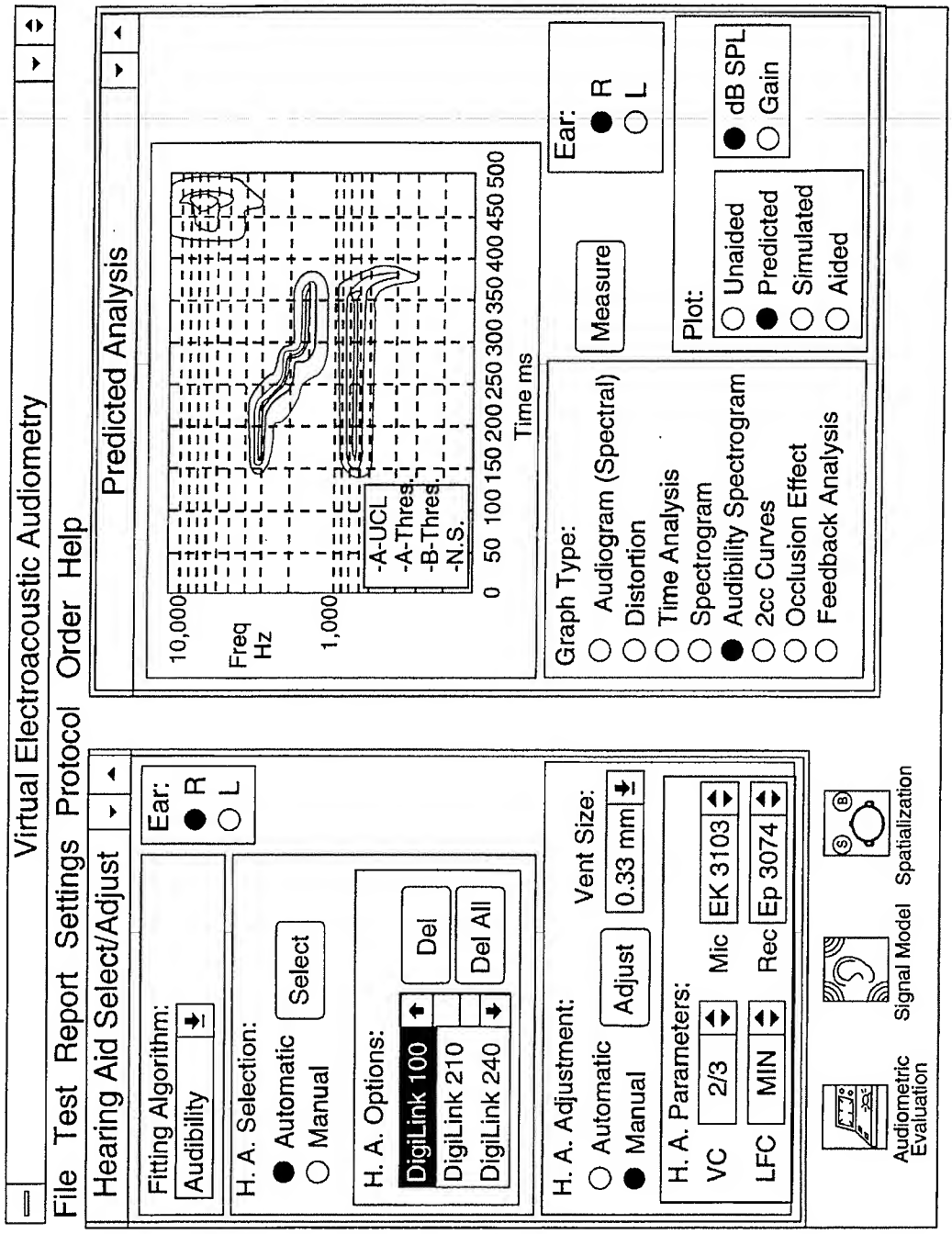
Teacher, female
S
65 dB SPL
Noise, Children
B
Audiometric Evaluation
Signal Model

Fig. 25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Hearing Aid Select/Adjust
Fitting Algorithm:
Audibility
Ear:
R
L
H. A. Selection:
Automatic
Manual
Select
H. A. Options:
DigiLink 100
DigiLink 210
DigiLink 240
Del
Del All
H. A. Adjustment:
Automatic
Manual
Adjust
Vent Size:
0.33 mm
H. A. Parameters:
VC
2/3
LFC
MIN
Mic
EK 3103
Rec
Ep 3074
Audiometric Evaluation
Signal Model
Spatialization
Predicted Analysis
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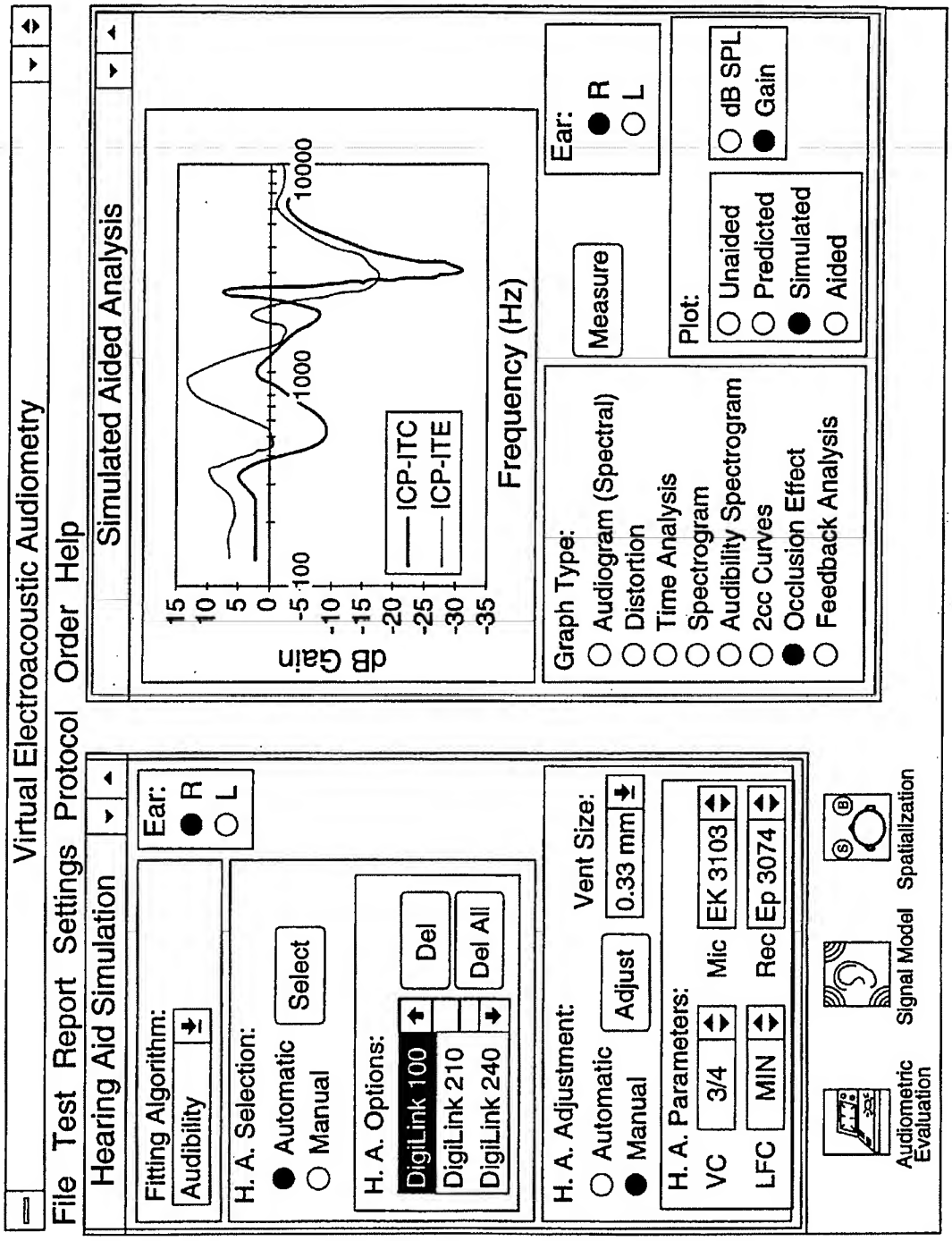
Freq Hz
1,000
-A-UCL
-A-Thres
-B-Thres
-N.S.
0 50 100 150 200 250 300 350 400 450 500
Time ms
Graph Type:
Audiogram (Spectral)
Distortion
Time Analysis
Spectrogram
Audibility Spectrogram
2cc Curves
Occlusion Effect
Feedback Analysis
Measure
Ear:
R
L
Plot:
Unaided
Predicted
Simulated
Aided
dB SPL
Gain

Fig. 26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Ear:
R
L
H. A. Selection:
Automatic
Manual
Select
H. A. Options:
DigiLink 100
DigiLink 210
DigiLink 240
Del
Del All
H. A. Adjustment:
Automatic
Manual
Adjust
Vent Size:
0.33 mm
H. A. Parameters:
VC
3/4
Mic
EK 3103
LFC
MIN
Rec
Ep 3074
Audiometric Evaluation
Signal Model
Spatialization
Simulated Aided Analysis
dB Gain
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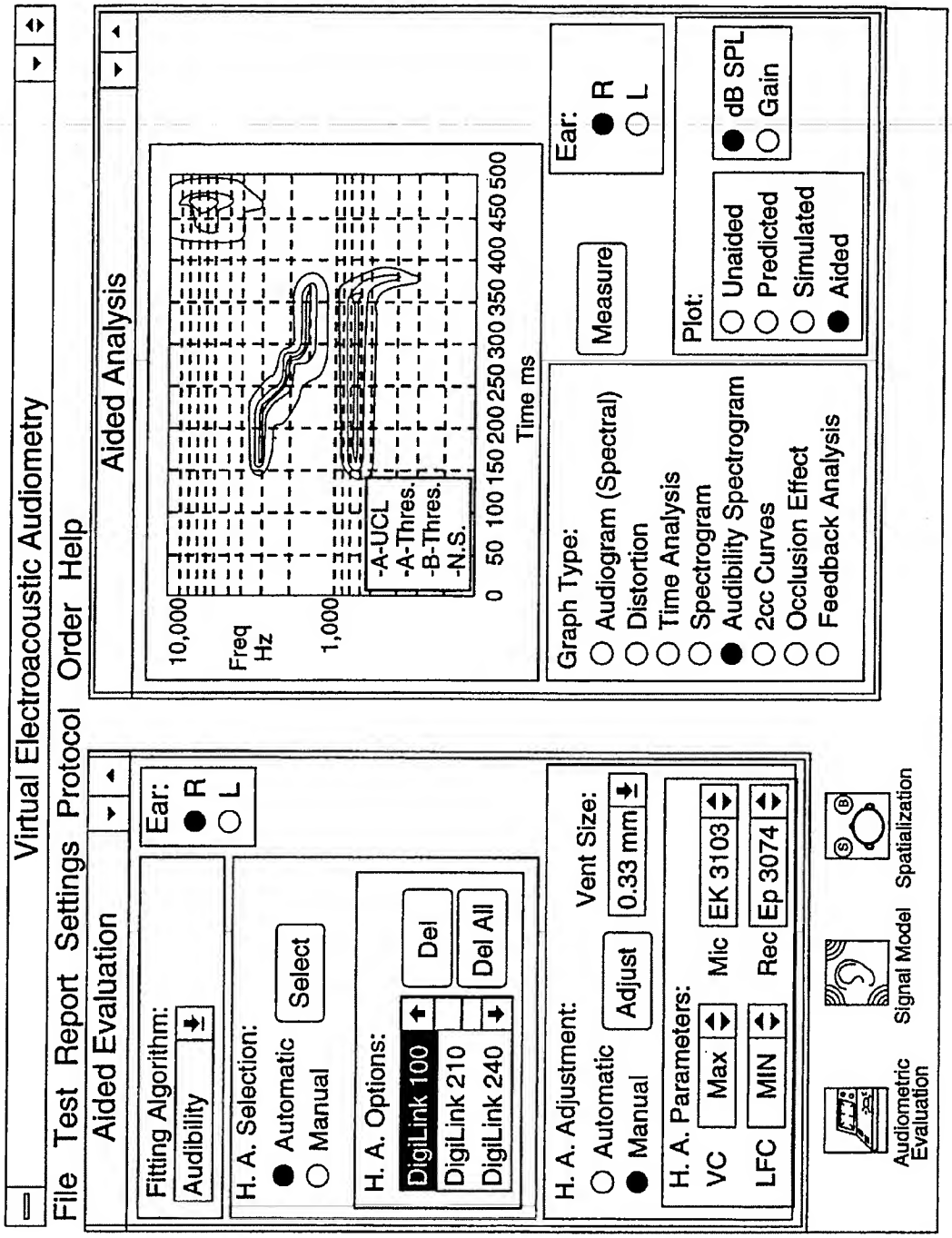
0
-5
-10
-15
-20
-25
-30
-35
100
1000
10000
ICP-ITC
ICP-ITE
Frequency (Hz)
Graph Type:
Audiogram (Spectral)
Distortion
Time Analysis
Spectrogram
Audibility Spectrogram
2cc Curves
Occlusion Effect
Feedback Analysis
Measure
Ear:
R
L
Plot:
Unaided
Predicted
Simulated
Aided
dB SPL
Gain

Fig. 27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R
L
H. A. Selection:
Automatic
Manual
Select
H. A. Options:
DigiLink 100
DigiLink 210
DigiLink 240
Del
Del All
H. A. Adjustment:
Automatic
Manual
Adjust
Vent Size:
0.33 mm
H. A. Parameters:
VC
Max
MIN
LFC
Mic
EK 3103
Rec
Ep 3074
Audiometric Evaluation
Signal Model
Spatialization
Aided Analysis
10,000
Freq Hz
1,000
-A-UCL
-A-Thres.
-B-Thres.
-N.S.
0 50 100 150 200 250 300 350 400 450 500
Time ms
Graph Type:
Audiogram (Spectral)
Distortion
Time Analysis
Spectrogram
Audibility Spectrogram
2cc Curves
Occlusion Effect
Feedback Analysis
Measure
Ear:
R
L
Plot:
Unaided
Predicted
Simulated
Aided
dB SPL
Gain

Fig. 28

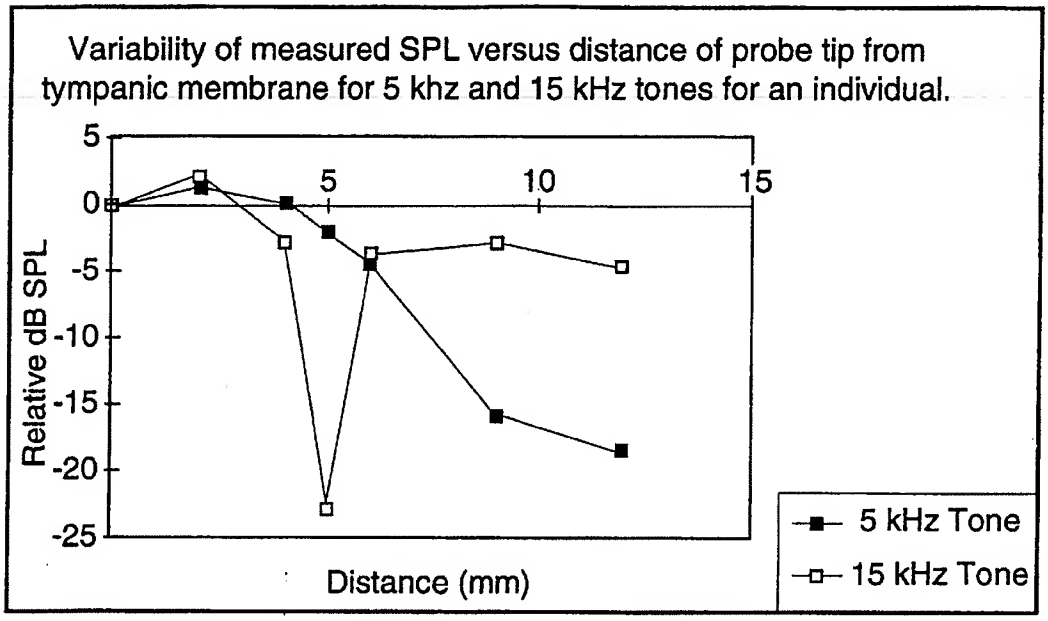
Variability of measured SPL versus distance of probe tip from tympanic membrane for 5 khz and 15 kHz tones for an individual.
Relative dB SPL
5
0
-5
-10
-15
-20
-25
5
10
15
Distance (mm)
5 kHz Tone
15 kHz Tone

Fig. 29

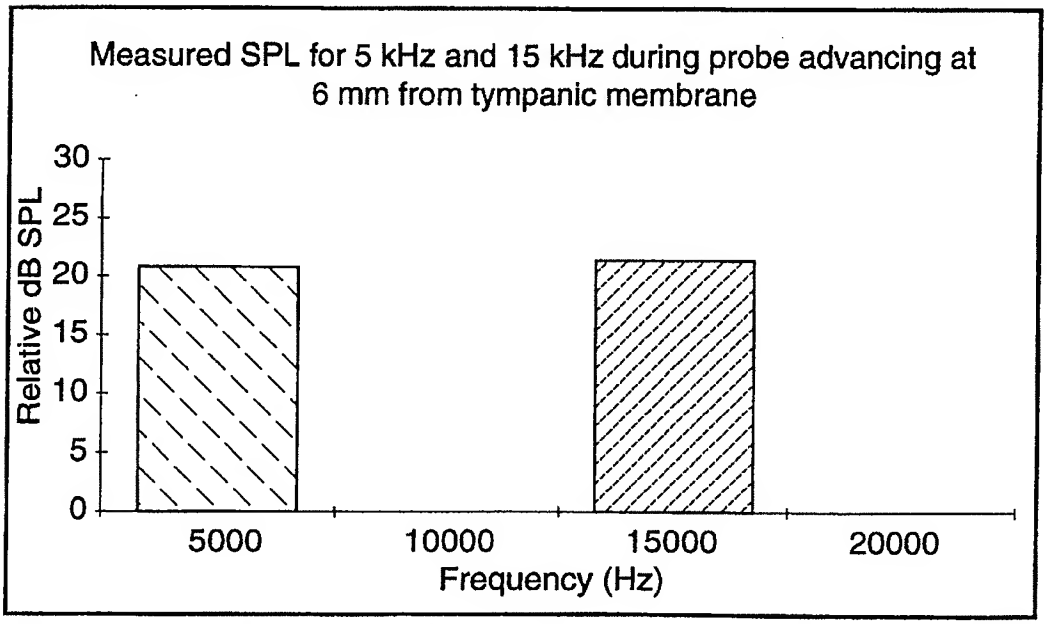
Measured SPL for 5 kHz and 15 kHz during probe advancing at 6 mm from tympanic membrane
Relative dB SPL
30
25
20
15
10
5
0
5000
10000
15000
20000
Frequency (Hz)

Fig. 30

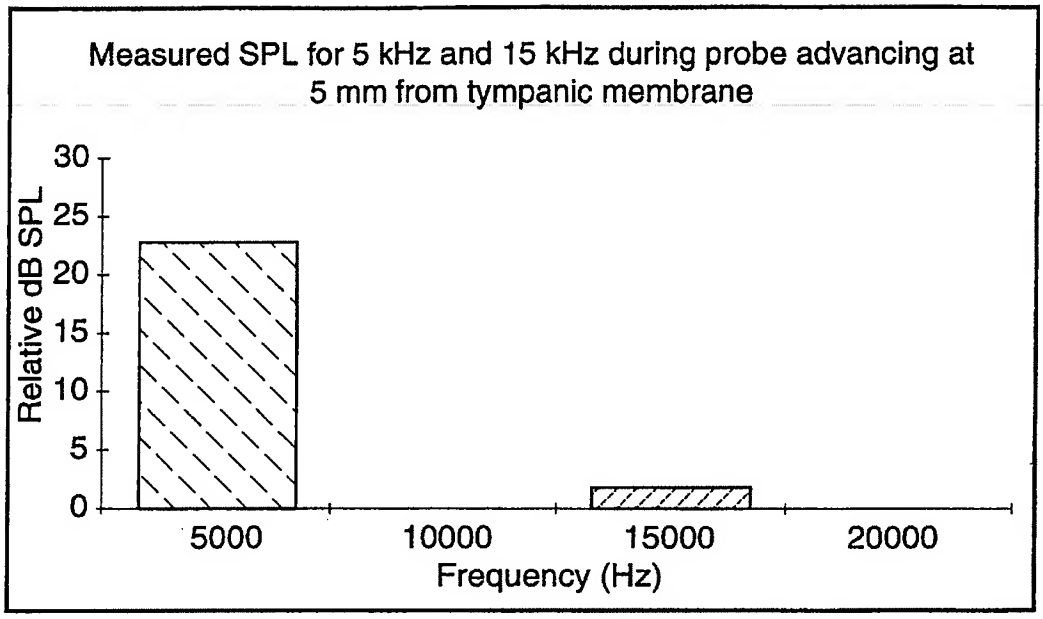
Measured SPL for 5 kHz and 15 kHz during probe advancing at
5 mm from tympanic membrane
Relative dB SPL
30
25
20
15
10
5
0
5000
10000
15000
20000
Frequency (Hz)

Fig. 31

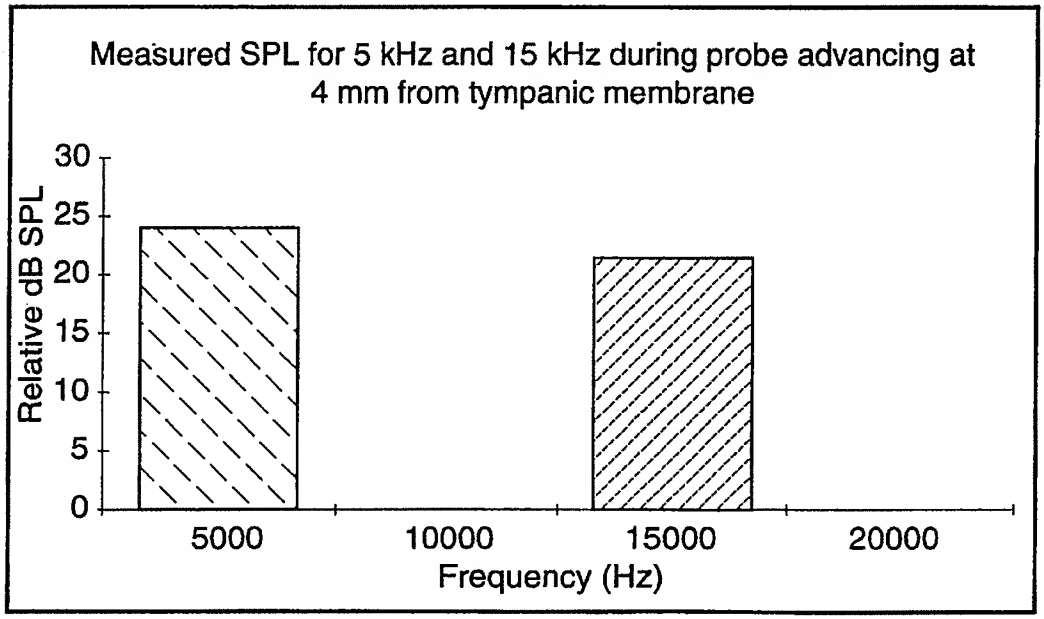
Measured SPL for 5 kHz and 15 kHz during probe advancing at
4 mm from tympanic membrane
Relative dB SPL
30
25
20
15
10
5
0
5000
10000
15000
20000
Frequency (Hz)

Fig. 32

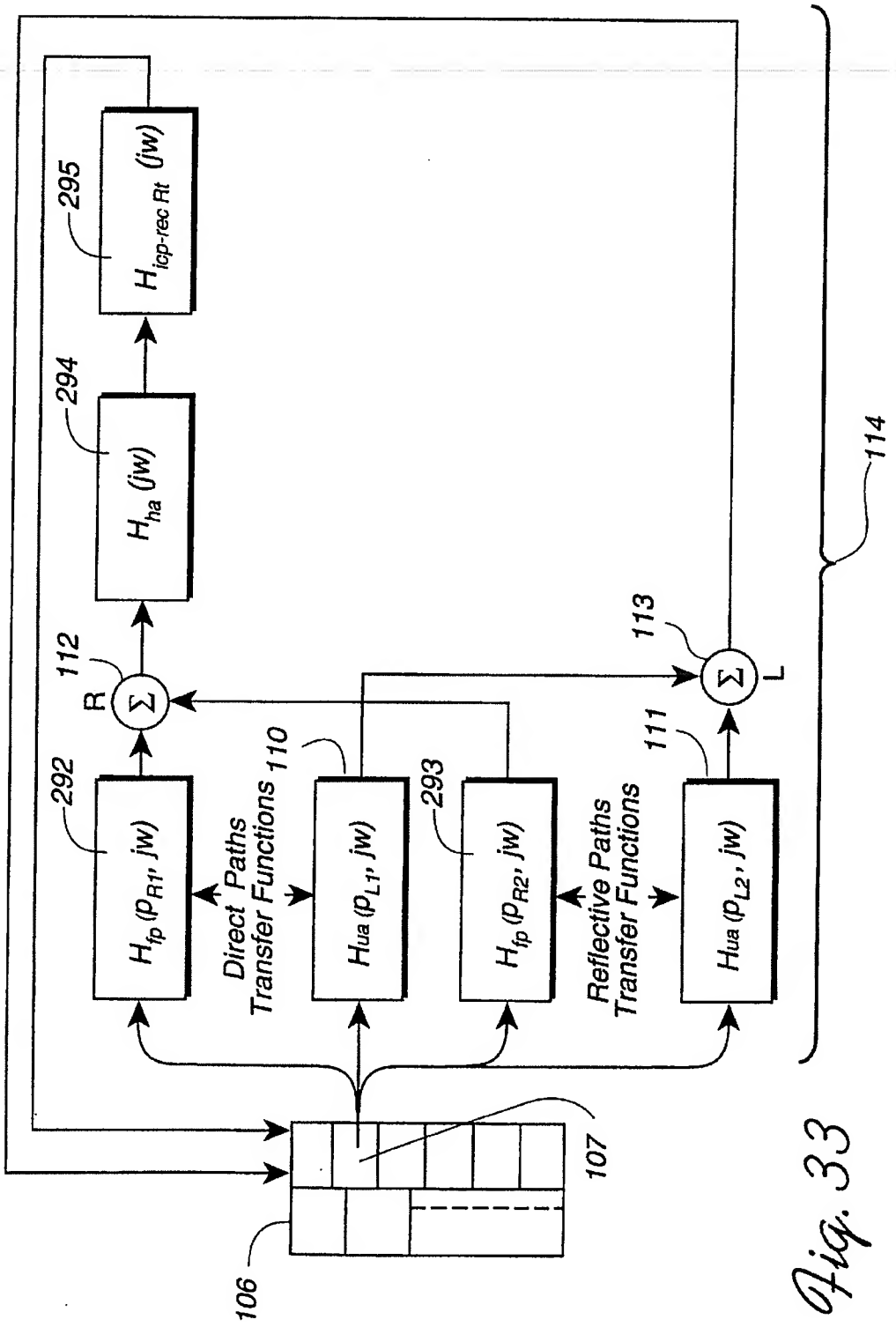
295
$H_{icp\text{-}rec\ Rt}$ (jw)
294
H_{ha} (jw)
112
R
Σ
113
Σ
L
114
292
H_{fp} (p_{R1}, jw)
Direct Paths Transfer Functions
110
H_{ua} (p_{L1}, jw)
293
H_{fp} (p_{R2}, jw)
Reflective Paths Transfer Functions
111
H_{ua} (p_{L2}, jw)
107
106

Fig. 33

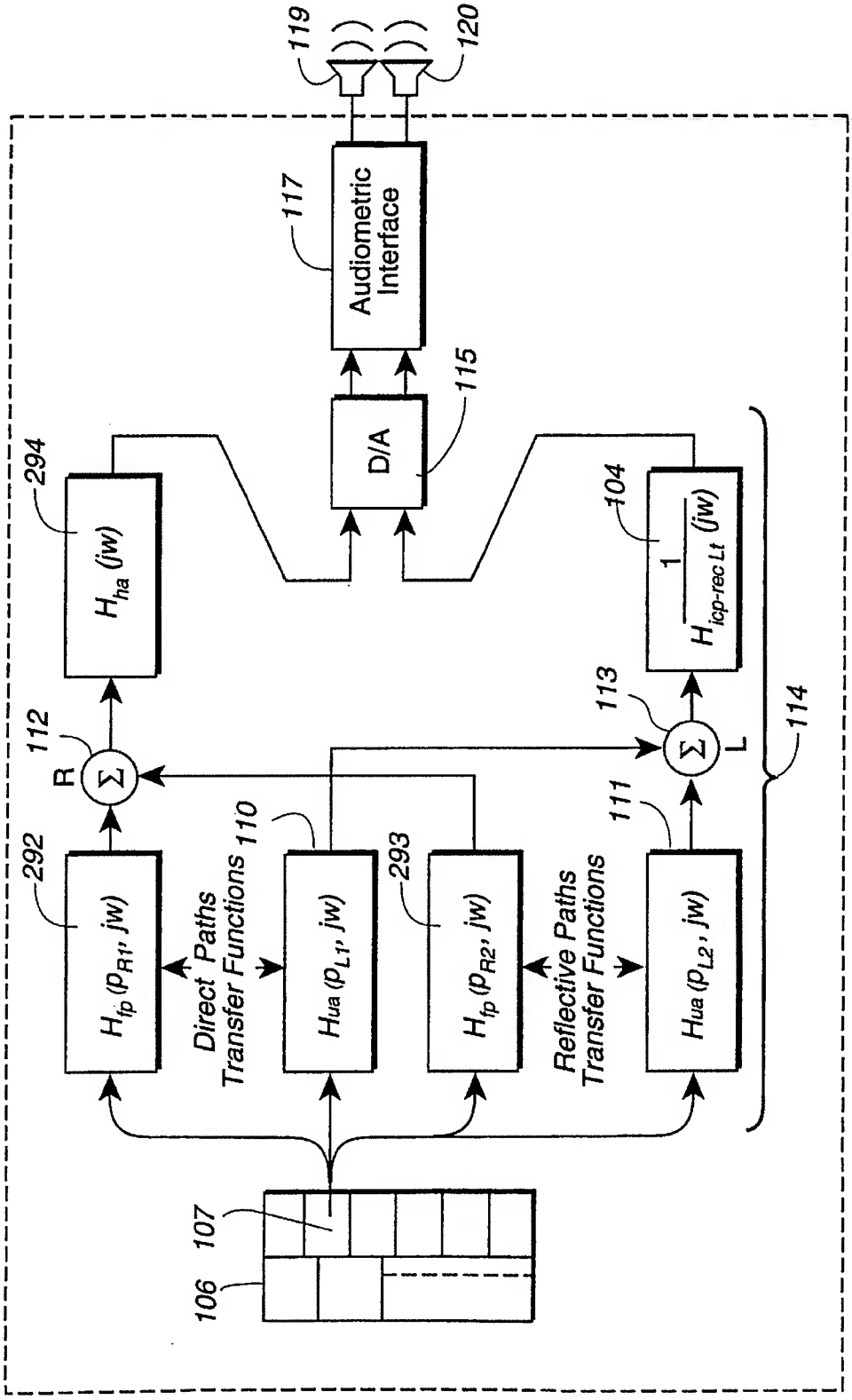
119
120
117
Audiometric Interface
115
D/A
294
H_{ha} (jw)
104
$\frac{1}{H_{icp\text{-}rec\ Lt}}$ (jw)
112
R
Σ
113
Σ
L
114
292
H_{fp} (p_{R1}, jw)
Direct Paths Transfer Functions
110
H_{ua} (p_{L1}, jw)
293
H_{fp} (p_{R2}, jw)
Reflective Paths Transfer Functions
111
H_{ua} (p_{L2}, jw)
107
106

Fig. 34

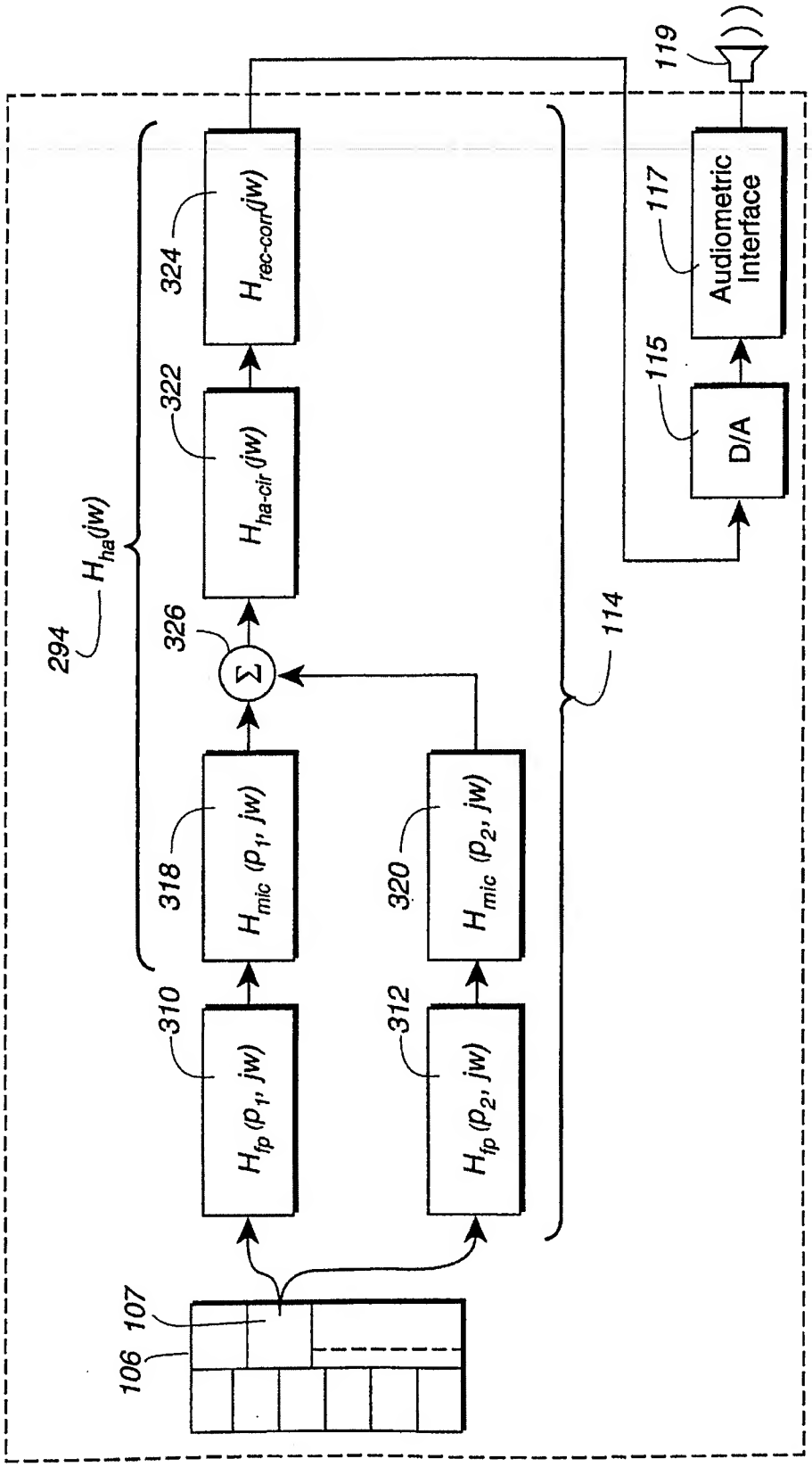
119
117
Audiometric Interface
115
D/A
324
$H_{rec\text{-}corr}(jw)$
322
$H_{ha\text{-}cir}(jw)$
326
Σ
294
$H_{ha}(jw)$
318
$H_{mic}(p_1, jw)$
320
$H_{mic}(p_2, jw)$
310
$H_{fp}(p_1, jw)$
312
$H_{fp}(p_2, jw)$
114
107
106

Fig. 35

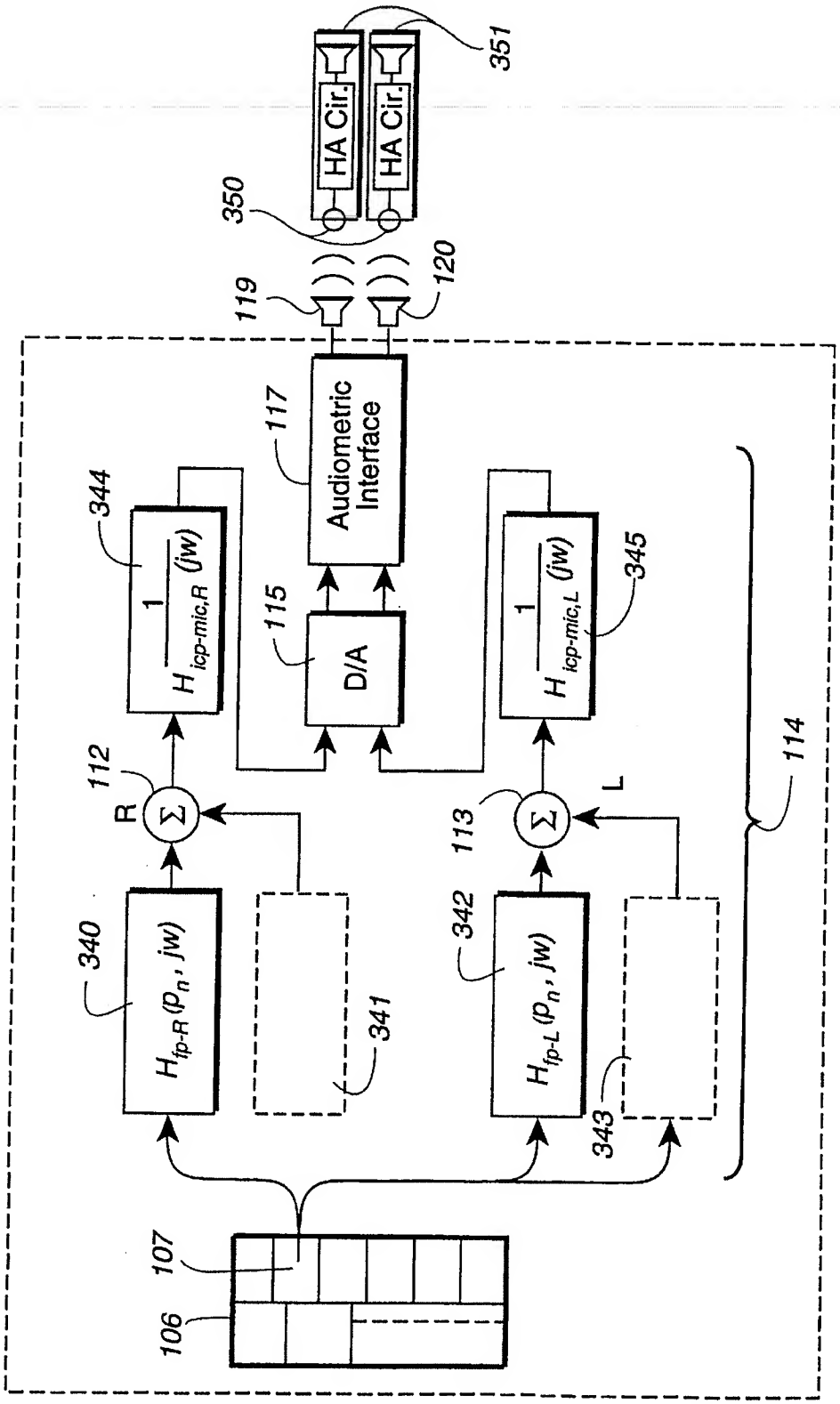
351
HA Cir.
HA Cir.
350
120
119
117
Audiometric Interface
344
345
115
D/A
112
R
113
L
114
340
341
342
343
107
106

Fig. 36

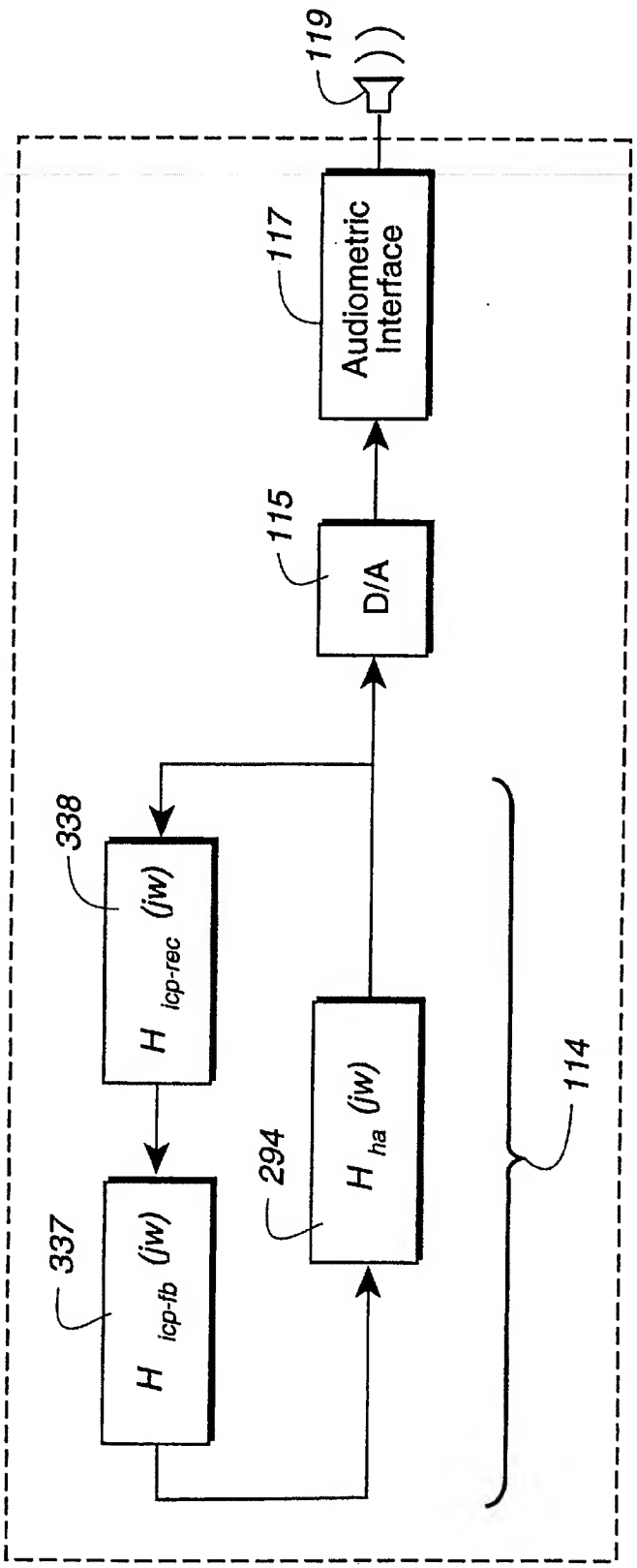
119
117
Audiometric Interface
115
D/A
338
$H_{icp\text{-}rec}(jw)$
337
$H_{icp\text{-}fb}(jw)$
294
$H_{ha}(jw)$
114

Fig. 37

SPATIALIZATION FOR HEARING EVALUATION

This is a divisional of copending application Ser. No. 08/292,073 filed on Aug. 17, 1994.

BACKGROUND OF THE INVENTION

1. Technical Field

The present invention relates to hearing evaluation and hearing aid fitting. More particularly, the present invention relates to virtual electroacoustic audiometry for unaided, simulated aided, and aided hearing evaluation.

2. Description of the Prior Art

The human auditory system processes sounds from a complex three-dimensional space via the external, middle, and inner ear, as well as via the complex neural pathways that lead to the auditory cortex within the brain. A measurable hearing loss, due to various conductive, sensorineural, or central auditory disorders, affects a significant percentage of the human population, particularly elderly persons. Rehabilitation via hearing aids remains the only viable option for those types of hearing impairments that cannot otherwise be medically treated or surgically alleviated.

Advances in hearing aids and fitting technologies are continuously being made. Today's ear-level hearing aids, i.e. in-the-ear (ITE), behind-the-ear (BTE), in-the-canal (ITC), and completely-in-the-canal (CIC) types, are more cosmetically appealing due to improvements in electronic and mechanical miniaturization. More significant, however, is the increasing availability of advanced hearing aid signal processing schemes, such as adaptive filtering and multiband dynamic compression.

As manufacturers are continuously developing new hearing aids with unique signal processing schemes, a hearing aid dispensing professional is faced with the increasingly difficult task of prescribing and selecting a hearing aid for a hearing-impaired individual from the available selection. A cursory look at available hearing aid processing schemes reveals an impressive array of categories, sub-categories, and associated acronyms that are baffling to most hearing aid dispensing professionals (see Mueller, H. G., *A Practical Guide To Today's Bonanza of Underused High-Tech Hearing Products,* The Hearing Journal, vol. 46, no. 3, pp. 13–27, 1993).

Today, optimal fitting of prescription hearing aids remains an elusive goal in auditory rehabilitation. The fundamental problem is that there are numerous electrical, acoustic, physical, and other parameters that affect hearing aid performance. These parameters include signal processing schemes, electronic circuit adjustments, size of hearing aid, insertion depth, venting size, patient controls, and life-style related factors that must be considered when prescribing and fitting a hearing aid. These hearing aid parameters are not only complex and highly interrelated, but also vary according to the unique interaction of the hearing device with the hearing-impaired individual.

Generally, the in situ performance characteristics of a hearing aid cannot be predicted with today's conventional fitting instrumentation and methods. Dissatisfaction among hearing aid user's, partially due to poor hearing aid prescription fitting, is manifested by a high return rates, often exceeding 20% according to industry reports.

Factors that Contribute to Unsatisfactory Hearing Aid Results

I. Inaccuracy of conventional diagnostic audiometry.

Assessment of hearing is the first step in the prescribing and fitting of a hearing aid. Accurate assessment of the individual's hearing function is important because all hearing aid prescriptive formulas depend on one or more sets of hearing diagnostic data (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* Singular Publishing Group, Inc., 1992: Ch. 5).

The hearing aid prescription process involves translating the diagnostic data into target hearing aid electroacoustic parameters that are used in the selection of the hearing aid. Traditional hearing evaluation methods and instruments employ a variety of air-conduction transducers for coupling acoustic signals into the ear. Commonly used transducers include supra-aural earphones, such as TDH-39, TDH-49, TDH-50, insert earphones, such as ER-3A, and free-field speakers (see *Specification of Audiometers,* ANSI-S3.6-1989, American Standards National Institute).

A threshold measurement obtained with such transducers is referenced to a mean threshold obtained by testing a group of otologically normal individuals. This mean threshold, by definition, is referred to as the zero decibel hearing-level or 0 dB HL. With this zero reference concept, threshold measurements of otologically normal persons can vary by 20 dB or more. These variations can be attributed to following factors:

1. Variability due to transducer type used and placement with respect to the ear.

In a study by Mowrer, et al discrepancies of 10 dB were found in 36% of threshold measurements (see Mowrer, D. E., Stearns, C., Threshold measurement variability among hearing aid dispensers, Hearing Instrument, vol. 43, No. 4, 1992). Another major disadvantage of measurements obtained using a traditional transducer is that results are not interchangeable with measurements taken with another transducer for a given individual (see Gauthier, E. A., Rapisadri, D. A., *A Threshold is a Threshold is a Threshold . . . or is it?,* Hearing Instruments, vol. 43, no. 3, 1992).

2. Variability due to transducer calibration methods that employ couplers that do not represent the human ear.

Although recently developed couplers more closely match the acoustic impedance characteristics of an average human ear, there is still disagreement as to the accuracy of this artificial ear (see Katz, J., *Handbook of Clinical Audiology,* Third Edition, 1985, pp. 126). Most calibration methods today rely on 6-cc or 2-cc couplers that are known to have considerable acoustic characteristic discrepancies from real human ears (see *Specification of Audiometers,* ANSI-S3.6-1989, American Standards National Institute). Furthermore, even if an agreement was made regarding an average artificial ear, variability among individuals is significant due to individual acoustic characteristics of pinna, ear canal, concha, and to a lesser extent, the head, and the torso (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 49–50). In one study, inter-subject variability was up to 38 dB across six standard audiometric frequencies when sound pressure levels (SPL) were measured at the tympanic membrane for 50 ears of 25 adults (see Valente, M., Potts, L., Valente, M., Vass, B., *Intersubject Variability of Real-Ear SPL:TDH-39P vs ER-3A Earphones,* In press, JASA).

3. Conventional audiometric measurement methods do not provide a means of self-calibration even though transducer characteristics are known for changes due to wear or damage of the moving diaphragm.

Clinicians who use regular subjective listening methods simply cannot detect gradual changes in transducer sensitivity.

Although errors due to the above factors are not likely to be accumulative in all cases, the potential for substantial errors is always present. Furthermore, these errors are not consistent across all frequencies and therefore cannot be simply compensated for during the fitting process via an overall volume adjustment.

II. Lack of realistic listening conditions in the unaided and aided hearing assessment.

1. Lack of Binaural Advantage Considerations.

Many studies have demonstrated the advantage of binaural versus monaural listening (see Cherry, E. C., *Some Experiments on the Recognition of Speech with One and Two Ears,* JASA, vol. 25, no. 5, 1953, pp. 975–979; Cherry, E. C., and Tylor, W, K., *Some Further Experiments on the Recognition of Speech with One and Two Ears,* JASA, vol. 26, 1954, pp. 549–554). These studies have focused on the advantages offered by the Binaural Masking Level Difference (BMLD) and Binaural Intelligibility Level Difference (BILD).

Early studies of BMLD and BILD involved the presentation of signal and noise to one or both ears at various phase relationships. Tone detection and speech intelligibility were shown to vary as much as 15 dB, depending on the signal/noise phase relationship. Even though many of these studies suggest the significance of binaural considerations, today's hearing assessment methods, unaided and aided, primarily deal with monaural test conditions, i.e. testing one ear at a time.

2. Lack of Spatialized Sound Considerations.

When audiometric signals such as speech and/or noise are delivered to the ear via a conventional audiometers and associated transducers, the sound perception by the test subject is not localized to any particular point in space (see *Specification of Audiometers,* ANSI-S3.6-1989, American Standards National Institute). For example, in speech audiometry evaluation, the speech stimuli level is adjusted for one ear and speech noise level is separately adjusted in the opposite ear. The test subject perceives sounds to be within the head and localization is limited to left/right direction. This type of signal presentation and perception is referred to as intracranial and is unlike the way humans normally perceive natural sounds. Recent studies by Bronkhorst and Plomp, and Begault expanded on previous binaural interaction advantage studies by employing headphone localization techniques (see Bronkhorst, A. W., Plomp, R., *The Effects of Head-Induced Interaural Time and Level Differences on Speech Intelligibility in Noise,* Journal of the Acoustical Society of America, vol. 83, no. 4, 1988, pp. 1508–1516; Bronkhorst, A. W.; Plomp, R., *The Effects of Multiple Speech-like Maskers on Binaural Speech Recognition in Normal and Impaired Hearing,* Journal of the Acoustical Society of America, vol. vol. 92, no. 6, 1992, pp. 3132–3139; and Bagault, D. R., *Call Sign Intelligibility Improvement Using a Spatial Auditory Display,* Ames Research Center, NASA Technical Memorandum 104014, April 1993). The results of these studies conclude the speech perception is not only dependent on intensity levels but also on the spatial relationship between speech and noise.

3. Lack of Evaluation Methods in Realistic Listening Environments.

Speech intelligibility and discrimination deteriorates in the presence of competing speech and other environmental sounds. Furthermore, the acoustic properties of a room, e.g. its walls and objects within the room, all play an important role in the filtering process subjected to the original signal source. These filtering effects are especially significant for hearing-impaired individuals who typically have a limited frequency response and dynamic range in their hearing function.

Today's methods of presenting competing and environmental sounds via conventional transducers fail to represent the acoustic reality of the typical listening condition. Recorded sound material presented via tape players, compact disks, or computer digital playback are subject to filtering effects of the transducer employed and/or the room acoustics of the clinical setup. There are no hearing assessment methods today that can evaluate or predict the hearing performance of an individual in a specific and realistic listening scenario.

For example, the hearing performance of a hearing-impaired child in a typical classroom in the unaided condition, and the hearing performance of the child with a specific hearing aid, i.e. aided hearing, in the same classroom environment. These and other auditory experiences are presently considered a fact of life that can not be dealt with in a clinical setup (see Mueller, H. G., Hawkins, D. B., Northern, J. L, *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 69).

III. Limitations of current real-ear measurement (REM) equipment and methods.

In recent years, real ear measurement (REM) systems were developed to assess the in situ performance of a hearing aid. REM consists of test probe measurements of the ear response to free field stimulus, i.e. speakers, taken at the tympanic membrane. A secondary reference microphone is typically placed outside the ear canal close to the ear canal opening. The reference microphone is used to calibrate the test probe as well as to regulate the stimulus level as the head moves with respect to the free field speaker.

For a comprehensive REM evaluation, measurement of the real ear response for the unaided, i.e. open canal, condition is first taken. Target hearing aid characteristics are then calculated based on the natural ear canal response characteristics, as well as other criteria (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, Ch. 5). When the hearing aid is prescribed, ordered, and received during a subsequent visit, the aid is inserted over the probe tube and adjusted to match the prescribed target hearing aid characteristics.

REM evaluation and REM-based prescriptive methods provide considerable improvements over previous fitting methods which relied on the combination of audiometric data and hearing aid 2-cc coupler specifications. Although REM offers insight into the in situ performance of the hearing aid, it suffers from several fundamental problems, as described below:

1. REM test results vary considerably depending on speaker position/orientation with respect to the ear, particularly at higher frequencies (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 72–74).
2. Real ear measurements are taken with a specific stimulus type, source-ear distance/orientation, and room acoustics. The specific test condition may not represent realistic listening scenarios encountered by hearing aid users. In fact, using conventional REM approaches, a hearing aid may be optimized for a specific listening condition while compromising the performance under other conditions that may be more important to the hearing-impaired individual.
3. Accurate REMs require careful placement of the test probe within the ear canal of an individual. The closer the probe to the tympanic membrane, the more accurate the results are, particularly for high frequency mea-

surements (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 74–79).

Present methods of probe placement are highly dependent on the operating clinician's skill and the specific length of the canal, which is about 25 mm for the average adult. Today's REM methods rely on visual observation of the probe tip.

This is especially problematic when a hearing aid is placed in the canal during the aided evaluation process. The only exception to the conventional visual method is the acoustic response method developed by Nicolet Corp. for use in the Aurora system (see Chan, J., Geisler, C., *Estimation of Eardrum Acoustic Pressure and Ear Canal Length from Remote Points in the Canal,* J. Acoust. Soc. Am. 87 (3), March 1990, pp. 1237–1247; and U.S. Pat. No. 4,809,708, *Method and Apparatus for Real Ear Measurements,* March 1989). However, Nicolet's acoustic response method requires two calibration measurements prior to placement of the probe at the desired position within the ear canal.

4. REM test results vary considerably depending on the placement of the reference microphone near the ear. The errors are especially significant at frequencies of 6 kHz and higher (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 72–74).
5. REM instruments employ sound field speakers in a room with ambient background noise that often exceeds 50 dB SPL across standard audiometric frequencies. This necessitates stimulus levels of 60 dB or higher to produce measurements having sufficient signal-to-noise ratios. This is problematic if hearing aid performance characterization under low level acoustic stimuli is required.

IV. The problem of correlating diagnostic, prescription formulae, and real ear measurements.

A significant factor that contributes to the results of a hearing aid fitting is the problem of adequately correlating diagnostic data with fitting needs of the hearing-impaired individual. Diagnostic measurements are typically taken in dB HL with transducers that are calibrated in 6-cc couplers. Hearing aid specification and performance measurements employ 2-cc couplers which do not represent the real-ear. Fitting involves the use of one of several prescriptive formulae, with results that are known to vary as much as 15 dB for the same diagnostic data across standard audiometric frequencies (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, p 107). These fitting formulae incorporate statistically based conversion factors that simplify the correlation of hearing aid requirements to a particular hearing impairment. However, averaged conversion factors are known to vary considerably with respect to objectively measured individual conversion factors.

Several methods and protocols have been suggested to alleviate errors associated with measurement errors and data correlation (see Sandberg, R., McSpaden, J., Allen, D., *Real Measurement from Real Ear Equipment. Hearing Instruments,* Vol. 42, No. 3, 1991, pp. 17–18). However, many of these protocols have not yet been widely accepted due to limitations of conventional audiometry and Real-Ear Measurement (REM) equipment and other factors related to efficiency of the proposed protocols in clinical setups.

Hearing rehabilitation through the use of hearing aids remains the only viable option for many hearing impaired individuals who cannot be medically or otherwise treated. A full audiometric evaluation is a required first step prior to fitting a hearing aid. Pure tones and one or more speech perception tests are typically involved in the basic audiometric test battery. Suprathreshold measurements may also be taken to establish a hearing dynamic range profile, in addition to the frequency response profile obtained in the threshold audiogram test. Following the audiometric evaluation, a hearing aid is then prescribed, selected, ordered, and subsequently tried and adjusted after being received from the manufacturer or assembled in the clinic. The fitting or determination of the electroacoustic parameters of a hearing aid typically involve a combination of objective measurements to achieve a desired target characteristics based on one of many prescriptive formulae and subjective measures based on the individual's subjective response to speech and other sounds at various loudness levels.

Conventional audiometry methods, employing headphones, inserts, or sound-field speakers, rely on presenting acoustic energy to the ear of the individual in a manner which is not representative of sound delivery under realistic listening conditions. Conventional audiometers present various tones, speech, and noise stimuli to each ear individually and thus are not capable of investigating the individual's binaural integration advantage, or of assessing the hearing function in a three-dimensional sound environment.

Another major disadvantage of conventional audiometry methods is the inability of such methods to assess accurately and objectively, in absolute physical terms such as dB SPL, the hearing function of an individual with respect to the inside of the ear canal to correlate unaided evaluation results to hearing aid requirements. One exception is the probe-mike-calibrated fitting system developed by Ensoniq, which only addresses testing accuracy (see Gauthier, E. A., Rapisadri, D. A., *A Threshold is a Threshold is a Threshold . . . or is it?: Hearing Instruments,* vol. 43, no. 3, 1992).

Furthermore, conventional audiometry instruments and methods are not capable of simulating the electroacoustic performance of one or more prescribed hearing aids and assessing their simulated function in realistic acoustic conditions relevant to the individual's unique listening requirements.

The master hearing aid concept, which gained some popularity in the '70s and '80s, involves an instrument that presents simulated hearing aids to the hearing aid user (see *Selection Instrumentation/Master Hearing Aids in Review, Hearing Instruments,* Vol. 39, No. 3, 1988). Veroba et al (U.S. Pat. No. 4,759,070, *Patient Controlled Master Hearing Aid,* Jul. 19, 1988) describe a patient controlled hearing aid module that is inserted into the ear canal and connected to a test module which offers multiple signal processing options, e.g. analog circuit blocks, to the individual. Hearing aid characteristics are determined by a tournament process of elimination, while the hearing-impaired person is presented with real-word sounds played back from tape decks via a set of speakers located around the hearing-impaired person's head. The system's fitting process is based on subjective responses of the hearing-impaired who must continuously decide on an alternative signal processing option, and supposedly eventually arrive at an optimal fitting.

The fitting process via the Veroba system, commercially known as the Programmable Auditory Comparator, an essentially obsolete product, does not involve any objective measurements or calculations for selecting and fitting of the

hearing aid. In fact, the entire fitting process is based on the subjective response of the hearing impaired person. Clearly, most hearing impaired individuals, on their own, cannot explore in a timely and efficient manner the spectrum of various complex and interrelated electroacoustic parameters of a hearing aid under various listening environments. A serious limitation of Veroba is that it does not teach how to assess objectively the performance of the simulated hearing aid, nor does it teach how the aided performance is related to the individual's unaided response determined previously during the audiometric evaluation process.

A major unsubstantiated claim in Veroba's system is the simulation of a realistic acoustical environment via tape-deck playback and speakers located around the head of the hearing-impaired individual. However, recorded acoustic signals that are played back are further subjected to acoustic modifications due to speaker characteristics, speaker position with respect to ear/head, and acoustic characteristics of the room, i.e. wall reflections and acoustic absorption. Without factoring in all of the specific acoustic modifiers in the transmission channel between the tape-deck and the individual's ear, a realistic listening condition cannot be achieved with Veroba or any such system. Furthermore, Veroba is not capable of manipulating the acoustic condition from its recorded form, e.g. by projecting an audio source in a specific location within a three-dimensional acoustic space with a specific acoustic boundary condition.

Another hearing aid simulator, the ITS-hearing aid simulator developed by Breakthrough, Inc. offers computer digital audio playback of digital recordings obtained from the output of various hearing aids (see *ITS-Hearing Aid Simulator,* Product brochure, Breakthrough, Inc., 1993). Each recording segment represents a specific acoustic input, listening scenario, hearing aid model, and hearing aid electroacoustic setting. The recording segments require memory space either on a hard disk or other known forms of memory storage devices, such as compact-disk read-only-memory. This digital-recording-based approach renders impractical the arbitrary selection of a hearing aid, hearing aid setting, and input stimulus for a hearing-impaired individual, when considering all the possible combinations. Furthermore, the effects of hearing aid vent sizes, and associated occlusion effect, insertion depth, and individual external ears, cannot be simulated with the proposed hearing aid simulator because it relies on conventional transducers, i.e. headphones and insert earphones.

For similar reasons, many other commercially available master hearing aid systems, do not have the ability to simulate accurately a hearing aid in a realistic listening environment. Furthermore, these systems do not include objective measurement methods for evaluating simulated aided versus unaided conditions. For these and other reasons, virtually all dispensed hearing aids today are fitted without the use of master hearing aid or hearing aid simulator instruments.

State-of-the-art REM equipment allows for in-the-ear-canal acoustic response measurements. The acoustic stimuli are typically generated by the REM equipment itself and delivered via a speaker, typically positioned at 0° azimuth, or with two speakers positioned at 45° azimuth, with the respect to the transverse plane of the head. The response measurements, i.e. free-field to real-ear transfer function, are essentially one-dimensional since they only provide a single transfer function per ear in a particular speaker-ear relationship, and are thus not capable of establishing a multi-dimensional profile of the real-ear response. Another disadvantage of conventional REM equipment and methods is the lack of real speech stimuli presentation because most REM equipment only offer pure-tone, pure-tone sweep, speech-noise and other speech-like stimuli. These stimuli do not explore responses to particular speech segments that may be important to the hearing-impaired individual during unaided and aided conditions.

Recent developments relating to electroacoustic hearing aid measures involve the testing of hearing aids in more realistic conditions. Real speech signals instead of pure tones and speech-like noise signals were employed in a recommended test protocol; and spectrogram plots indicating temporal, i.e. time, analysis of the acoustic energy in dB SPL versus frequency was compared for hearing aid input versus output (see Jamieson, D., *Consumer-Based Electroacoustic Hearing Aid Measures,* JSLPA Suppl. 1, Jan. 1993). The limitations of the proposed protocol include: limited acoustic reality due to the specified sound delivery method via a speaker to a hearing aid in an enclosed chamber; and limited value of the spectrogram plots which do not directly indicate the relationship of the plot to audibility and loudness discomfort.

Other recent developments involve three-dimensional sound presentation via headphone transducers (see Wightman, F. L., Kistler, D. J., *Headphone Simulation of Free-Field Listening. I: Stimulus Synthesis,* JASA. vol. 85, no. 2, 1989, pp. 858–867; and Wightman, F. L., Kistler, D. J., *Headphone Simulation of Free-Field Listening. II: Psychophysical Validation,* JASA. vol. 85, no. 2, 1989, pp. 868–878). These three-dimensional effects are achieved by recreating the in-the-ear-canal acoustic response to free-field signals via headphones or speakers (see U.S. Pat. No. 4,118,599, *Stereophonic Sound Reproduction System,* Oct. 3, 1978; U.S. Pat. No. 4,219,696, *Sound Image Localization Control System,* Aug. 26, 1980; U.S. Pat. No. 5,173,944, *Head Related Transfer Function Pseudo-Stereophony,* Dec. 22, 1992; U.S. Pat. No. 4,139,728, *Signal Processing Circuit,* Feb. 13, 1979; and U.S. Pat. No. 4,774,515, *Altitude Indicator,* Sep. 27, 1988). This involves digital filtering of source signals based on head-related-transfer-function (HRTF). The HRTF, essentially real-ear unaided response (REUR) in three-dimensional space, is a frequency dependent amplitude and time delay measurement that results from head shadowing, pinna, concha, and ear canals. The HRTF enables externalization of localized sound with headphones. Source signals that are processed with HRTF provide the listener with free-field listening experience according to the controls of the signal processing parameters.

Present research and development efforts in three-dimensional audio is mainly focused on commercial musical recordings, playback enhancement, and human-machine interface enhancement (see Bagault, D. R., *Call Sign Intelligibility Improvement Using a Spatial Auditory Disaply,* Ames Research Center, NASA Technical Memorandum 104014, April 1993; and Begault, D., Wenzel, E., *Headphone Localization of Speech,* Human Factors, 25 (2), pp. 361–376, 1993) and virtual reality systems (see *The Beachtron-Three-dimensional audio for PC-compatibles,* reference manual, Crystal River Engineering, Inc., Revision D, November, 1993). The object of these three-dimensional audio systems has been limited to simulating situational awareness in an approximate virtual acoustic environment since non-individualized HRTF set is typically employed.

The application of three-dimensional audio in objective in-the-ear-canal assessment of hearing in the unaided, simulated aided, and aided conditions would be a significant and extremely helpful departure from known audiometric techniques.

SUMMARY OF THE INVENTION

The invention provides a virtual electroacoustic audiometer (VEA), which is a system used in the assessment of human hearing function in the unaided, simulated aided, and aided conditions. A pair of intra-canal prostheses (ICP) are placed in the two ear canals of an individual to deliver acoustic stimuli. A probe measurement system, partially inserted in the ICP, measures the in-the-ear-canal response conditions near the tympanic membrane during all hearing evaluation, thus providing a common reference point for correlating responses in the unaided, simulated aided, and aided evaluation conditions. A unique modular hearing aid defined in accordance with the results of such hearing assessment is also provided that includes highly configurable electroacoustic and electronic signal processing elements.

During unaided evaluation, the system performs audiometric tests, such as pure tone thresholds, uncomfortable loudness levels (UCL), speech reception threshold, and speech discrimination. These peripheral hearing tests, as well as other central auditory processing (CAP) tests, evaluate the hearing function of the human in response to acoustic stimuli measured near the tympanic membrane in absolute sound pressure level (SPL) terms, unlike conventional stimuli which are presented in relative hearing level (HL) terms.

Another significant feature of the VEA is its ability to synthesize, or create, acoustic signals that are representative of signals received in real listening environments in a three-dimensional space. This is achieved by incorporating the various filtering effects of room acoustics, atmospheric absorption, spreading loss, interaural delay, and spectral shaping of external ear, and other body effects. For example, a listening condition representing a teacher-talker in classroom is digitally synthesized and acoustically delivered via the ICP to a child to assess his/her unaided and aided listening ability in a classroom environment. Spatialized competing signals representing school children noise is optionally presented in addition to the spatialized primary speech signal, i.e. the teacher, to assess further the child's speech discrimination ability in the presence of background noise.

The unaided evaluation method involves both ears in the listening experience similar to the way humans normally hear sounds, with each ear receiving a portion of the acoustic energy according to the relationship between each ear and the various virtual audio sources. In contrast, conventional audiometry methods present intracranial acoustic stimuli to each ear individually, for example, speech to one ear, and competing noise in the opposite ear.

The simulated aided assessment of the VEA system is accomplished by incorporating the electroacoustic performance of a desired hearing aid into the unaided digital synthesis of acoustic signals. The simulated hearing aid electroacoustic parameters include microphone and receiver transfer functions, and amplifier and filter characteristics.

Specific or generalized acoustic models are digitally presented to the input of the simulated hearing aid process. Specific acoustic models represent listening scenarios that are important to the individual under evaluation and that may be selected and manipulated by the operating clinician, for example a teacher-talker source model in a classroom environment model with a specific source-ear relationship. A typical goal in such a specific scenario is to maximize speech intelligibility by optimizing the electroacoustic characteristics of the simulated hearing aid. Generalized acoustic conditions represent listening scenarios that are associated with normative response data. An example of a generalized model is an audiologic word list, such as W-22, having a specific spatialized background noise. Test scores are compared with general model normative data stored in the system's memory.

The VEA system also simulates other hearing aid effects that can not be simulated by the digital synthesis process due to the unique effects of the individual ear. These include the occlusion effect, venting size, and oscillatory feedback potential. The occlusion effect is a phenomenon that results in changes to the perceived characteristics of the individual's own voice when the ear canal is occluded with a hearing aid.

In addition, the VEA system offers a method of measuring various individualized acoustic transfer functions in a three-dimensional space, which are incorporated during the various synthesis processes to create virtual acoustic conditions for an individual.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block level schematic diagram showing the major components of the VEA system, including dual ICP prostheses inserted in the ear canal of an individual; a probe microphone system; and a computer system including a digital audio synthesizer module, a digital audiometer module, and a virtual acoustic space measurement module according to the invention;

FIG. **2** is a block level schematic diagram of a digital audio synthesizer module according to the invention;

FIG. **3** is a block level schematic diagram of a digital audiometer module according to the invention;

FIG. 4 is a block level schematic diagram of a virtual acoustic space measurement module according to the invention;

FIG. **5** is a block level schematic diagram of a virtual acoustic space measurement system according to the invention;

FIG. **6** is a perspective view of an adjustable chair used for positioning a patient's head during virtual acoustic space testing;

FIG. **7** is a schematic diagram showing speaker arrangement in a virtual acoustic space measurement system, including transverse plane speakers, and sagittal plane speakers according to the invention;

FIG. **8** is a schematic diagram showing an example of transfer function interpolation at a point i_3 from transfer functions, measured at points m_1 and m_2 in a two-dimensional transverse plane according to the invention;

FIG. **9** is a schematic diagram showing an example of realization of a realistic listening scenario for unaided hearing evaluation conditions, and in particular showing a teacher-talker/child-listener scenario including direct acoustic paths P_{R1} and P_{L1} and early reflection paths P_{R2} and P_{L2} to the right and left ears of the child-listener according to the invention;

FIG. **10** is a block level schematic diagram showing an example of realization of a realistic listening scenario for unaided hearing evaluation conditions, and in particular showing a process representation of a teacher-talker/child-listener scenario during unaided evaluation according to the invention;

FIG. **11** is a partially sectioned, perspective view showing an intra-canal prosthesis (ICP) for an ICP-ITE representing hearing aids for shallow ear canal placement according to the invention;

FIG. **12** is a partially sectioned, perspective view showing an intra-canal prosthesis (ICP) for an ICP-ITC representing hearing aids for deep ear canal placement according to the invention;

FIG. **13** is a perspective view showing an intra-canal prosthesis (ICP) face-plate end, including face-plate probe tube holders and probe tube placement according to the invention;

FIG. **14** is a partially sectioned, side view showing an ICP core module for a two-part ICP configuration according to the invention;

FIG. **15** is a partially sectioned, side view showing adjustable vent inserts and an ICP-ITE sleeve for an ICP-ITE configuration according to the invention;

FIG. **16** is a partially sectioned, side view showing an ICP-ITC sleeve for a two-part ICP configuration according to the invention;

FIG. **17** is a partially sectioned, side view showing a complete two-part ICP-ITC assembly according to the invention;

FIG. **18** is a partially sectioned, side view showing an ICP having a programmable vent according to the invention;

FIG. **19** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including direct acoustic coupling via a magnetic attraction method according to the invention;

FIG. **20** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including direct acoustic coupling via an acoustic coupler method according to the invention;

FIG. **21** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including a programming and acoustic coupling interface according to the invention;

FIG. **22** is a partially sectioned, side view showing a hearing aid and acoustic coupling to an ICP via an acoustic coupler tip according to the invention;

FIG. **23** is a block level schematic diagram showing an example of a fitting process provided by the virtual electroacoustic audiometer system according to the invention;

FIG. **24** is a graphic computer generated display showing a reference measurements module according to the invention;

FIG. **25** is a graphic computer generated display showing an unaided evaluation module according to the invention;

FIG. **26** is a graphic computer generated display showing a predicted aided module according to the invention;

FIG. **27** is a graphic computer generated display showing a simulated aided evaluation module according to the invention;

FIG. **28** is a graphic computer generated display showing an aided evaluation module according to the invention;

FIG. **29** is a line graph plotting the variability of measured SPL versus distance of probe tip from tympanic membrane for 5 kHz and 15 kHz tones for an individual according to the invention;

FIG. **30** is a bar graph plotting the measured SPL for 5 kHz and 15 kHz during probe advancing at 6 mm from tympanic membrane according to the invention;

FIG. **31** is a bar graph plotting the measured SPL for 5 kHz and 15 kHz during probe advancing at 5 mm from tympanic membrane according to the invention;

FIG. **32** is a bar graph plotting the measured SPL for 5 kHz and 15 kHz during probe advancing at 4 mm from tympanic membrane according to the invention;

FIG. **33** is a block level schematic diagram showing an example of a teacher-talker/child-listener scenario using predicted aided evaluation for the right ear according to the invention;

FIG. **34** is a block level schematic diagram showing an example of a teacher-talker/child-listener scenario using simulated aided evaluation for the right ear according to the invention;

FIG. **35** is a block level schematic diagram showing a simulated hearing aid with directional microphone according to the invention;

FIG. **36** is a block level schematic diagram showing an example of the realization of realistic listening scenarios for aided hearing evaluation conditions according to the invention; and

FIG. **37** is a block level schematic diagram showing an example if the prediction and simulation of oscillatory feedback of a simulated hearing aid.

DETAILED DESCRIPTION OF THE INVENTION

For purposes of the description herein, the following definitions shall be consistently applied:

Window: Refers to a graphical area displayed on a computer screen, that represents a collection of controls, objects, entry fields, and plots, that are grouped together according to a logical functional manner.

Iconized: Refers to an active window that is shown as an icon. Its display is disabled but may be enabled by clicking on the icon on the computer screen.

The virtual electroacoustic audiometer (VEA) described herein is a unitary instrument that is used in the hearing assessment in the unaided, simulated aided, and aided conditions. The VEA also offers new methods for hearing aid fitting and analysis using a combination of digital synthesis of realistic acoustic stimuli and in-the-ear-canal response measurements throughout the assessment and fitting processes.

FIG. **1** shows the main components of the preferred embodiment of the VEA system **15**. A pair of intra-canal prostheses (ICP) **22** is inserted in the ear canal **21** of an individual for delivering acoustic stimuli **25** in a manner similar to that of a hearing aid. Each ICP contains a receiver, i.e. a speaker, for transmitting acoustic signals to the tympanic membrane **26**. The ICP also contains a probe tube **24** for measuring the acoustic response that results from the unique interaction of the receiver-produced acoustic stimuli and the ear-canal characteristics of the individual. A probe microphone system consisting of a probe tube **24** and probe microphone **23** measures acoustic signals from the ear canal **21** and provides electrical signals representative of the acoustic signals. A response keyboard **27** is provided to register a response from the test subject **20** during various hearing evaluation tests.

Each ICP receiver **22** is electrically connected to a digital audiometer module **19** that provides an interface to various audiometric transducers including the ICP receiver **22** and probe measurement system **23**. The digital audiometer module is connected to a digital audio synthesizer module **18** and a virtual acoustic space measurement module **14** via various inter-module cables. The virtual acoustic space measurement module includes an output terminal **16** for connection to a plurality of test speakers. These modules may be contained at or within a standard personal computer (PC) **11** that also contains standard computer accessories such as

memory storage devices **17**, a display monitor **10**, a keyboard **12**, and a mouse **13**. Memory storage devices are collectively referred to as system memory **17**.

Block diagrams of the digital audio synthesizer, digital audiometer and virtual acoustic space measurement modules are shown in FIGS. **2**, **3**, and **4**.

In the exemplary embodiment of the invention, the digital audio synthesizer, digital audiometer, and virtual acoustic space measurement modules are connected to the personal computer system via the Industry Standard Architecture (ISA)-bus interface **34** and ISA-bus **39** of the personal computer (see, for example FIG. **2**). Digital data representing audio sources are retrieved from the system memory via the bus interface **34**, and are digitally processed by a digital signal processor **33** within the digital audio synthesizer module **18**. The digitally processed data are then converted to analog form using an digital-to-analog converter **35** that typically operates at conversion rate of 44.1 kHz, or at another rate depending on the desired signal bandwidth required.

The digital audio synthesizer module also receives analog signals representing audio signals via its input connector **31** from external audio sources such as tape or CD players (not shown). Received analog signals are converted to digital signals by the analog-to-digital converter **32** for signal processing via digital signal processor **33**.

Multiple digital audio synthesizer modules (not shown) may be used to enhance the system's digital signal processing capability. This is particularly useful for parallel real-time binaural signal synthesis. Multiple digital audio synthesizer modules are cascaded by connecting the output **38** of one digital audio synthesizer module to the auxiliary input **30**, or input **31** of another digital audio synthesizer module. The internal and auxiliary signals are combined within the module at a summing node **36** prior to output. In the preferred embodiment of the invention, two digital audio synthesizer modules are used. Each module employs a Motorola DSP56001 digital signal processor clocked at 40 MHz.

The analog output **38** from the digital audio synthesizer module **18** is routed to the mixer **45** of the digital audiometer module **19** (FIG. **3**) via a connector **42**. Analog audio signals received at the digital audiometer module are mixed via mixer circuit **45**, amplified via an audio amplifier circuit **46**, and impedance matched and routed to various audiometric transducers via an audiometric transducer interface circuit **49**. Outputs to audiometric transducers include ICPs **50** (discussed above, and in further detail below), bone vibrators **51** (not shown), a headphone **52** (not shown), and other conventional methods of delivering sounds to the ear of an individual.

Amplified signals from the audio amplifier **46** are also sent to the digital audio synthesizer module input **31** from an audio buffer circuit **47** output connection **48**. The mixer circuit **45** also includes connections for receiving audio signals from ICP microphones **55**, an operating clinician microphone **56** (not shown), and a patient microphone **57** (not shown), via a microphone amplifier **58**.

External line-level signals received at input connectors **53** are also amplified via an amplifier **54** and sent to mixer circuit **45**. A response keypad interface circuit **60** is employed to interface the system to the response keypad via a connector **59** to register an individual's response to acoustic stimuli during various audiometric evaluation processes. The operating clinician microphone, connected to the digital audiometer module, allows the operating clinician to communicate with the patient via the ICP pair. The patient microphone allows the patient to communicate back to the operating clinician during certain audiometric tests that require verbal responses from the patient. The patient microphone is also used in occlusion effect measurements, as are described in more detail below.

The digital audiometer module also includes a PC-BUS connection **43** and PC-BUS interface circuit **44** that link the digital audiometer module to the VEA to coordinate module operation at the system level.

The VEA also includes a virtual acoustic space measurement system (FIG. **5**) that is used to evaluate the individual's acoustic transfer function set. A block diagram of the virtual acoustic space measurement module **14** is shown in FIG. **4**. The virtual acoustic space measurement module receives electrical signals, representing various acoustic signals, from the digital audio synthesizer module output connectors **38** via a set of input connectors **64**. Input signal level adjustment and routing is accomplished via a mixer circuit **65**, an audio amplifier circuit **66**, and a speaker routing and interface circuit **71**. The output of the virtual acoustic space measurement module is thence coupled to various test speakers in a speaker array **16**.

The virtual acoustic space measurement module also includes a PC-BUS connection **68** and PC-BUS interface circuit **67** that link the virtual acoustic space measurement module to the VEA to coordinate module operation at the system level. Such coordination includes processing information indicative of patient head position connected to the module from a patient head positioning sensor via a connector **70** and a positioning sensor interface circuit **69**.

An adjustable chair **78** is preferably used to ensure proper ear positioning within the measurement space, as shown in FIG. **6**. A vertical adjustment lever **79** adjusts the vertical position of the individual on the chair. A back adjustment knob **81** adjusts a chair back support **80**. The head support **82** is adjustable to support the head of the individual seated on the chair. An ear position reference arm **84** provides a target reference by pointing a set of ear canal opening pointers **83** to the individual's ear canal openings. The ear position reference arm **84** is preferably removable from the ear area via a reference arm vertical adjustment knob **85** to minimize acoustic reflections into the ear area during transfer function measurements.

An infrared tracking method (not shown) may also be used to position and maintain the head in the proper position with respect to the speaker array **16**, FIG. **5**; **89–94**, FIG. **7**). A light- reflective target object (not shown) placed just below the ear lobe of the individual, may be used to reflect the infrared light from the incident infrared light emitter. Proper ear placement is indicated by reflected light which is detected by the positioning sensor interface **69** (FIG. **4**).

The virtual acoustic space measurement system generates various sets of transfer functions that are used during the hearing evaluation process. Generally, a transfer function of a linear system defines a complex function H(jw) having magnitude and phase characteristics that are dependent on frequency (w). Once a transfer function H(jw) is determined, a system's response to an arbitrary input signal can be predicted or synthesized.

The transfer function set in the virtual acoustic space measurement system is obtained from a set of acoustic sources, such as speakers, positioned in a three-dimensional space. The preferred speaker setup is an array of six speakers **89–94** positioned at an equal distance (d) from a patient head reference point **88**, as shown in FIGS. **5** and **7**. The head

reference point **88** is defined as the point bisecting the line joining the centers of the openings of the ear canal **21**.

Four of the speakers, i.e. **#1 (89)**, **#2 (90)**, **#3 (91)**, and **#4 (92)** are located in the transverse plane **95** containing the head reference point **88**. Speakers **1** through **4** are positioned at azimuth angles 0°, 45°, 315°, and 270°, respectively, as shown in FIG. **7** at A. Three of the speakers, i.e. **#1 (89)**, **#5 (93)**, and **#6 (94)** are located in the sagittal plane **96** containing the head reference point **88**. Speakers **#1**, **#5**, and **#6** are positioned at altitude angles of 0°, 45°, and −45°, respectively, as shown in FIG. **7** at B.

A set of transfer functions for the six-speaker configuration shown in FIG. **7** allows six pairs, i.e. right and left ear measurements, of frontal measurements where the head is facing speaker **#1**. An additional six pairs of back measurements are preferably taken where the head is facing opposite (not shown) to speaker **#1**. Accordingly, a complete transfer function set consists of 12 pairs of measurements that represent finite points in a sphere of a radius (d). Of the twelve paired measurements, eight paired measurements are in the transverse plane and six paired measurements are in the sagittal plane. Two paired measurements are common to both planes. Paired measurements contain not only individual transfer functions for each ear, but also contain the interaural phase relationship with respect to each speaker.

A transfer function measurement set with a pair of probes placed near the tympanic membrane in the unoccluded ear canal is referred to herein as the unaided transfer function $H_{ua}(p_n, jw)$, where p_n is the location of speaker n defined by polar coordinates d, θ, and α, where d is the distance between the speaker and the head reference point as shown in FIG. **7** at A.; θ is the azimuth angle of sound incidence with the respect to transverse plane as shown in FIG. **7** at A.; and α is the altitude angle with respect to the sagittal plane as shown in FIG. **7** at B. $H_{ua}(p_n, jw)$ represents the acoustic transfer function that results from sound propagation from a speaker #n to the tympanic membrane when various acoustic factors are considered, including atmospheric propagation losses, effects of head, torso, neck, pinna, concha, ear canal, tympanic membrane, and middle ear impedance.

Transfer function measurements with a probe tube placed on the face-plate of the ICP may also be made. These measurements are referred to herein as $H_{fp}(p_n, jw)$, which represent the transfer function from a speaker #n to a face-plate (fp) of the ICP (discussed in more detail below), at a location representative of the microphone position on a face-plate of a simulated hearing aid.

Generally a transfer function $H(p(d,\theta,\alpha), jw)$ at an arbitrary point p_d, θ,α in space at coordinates d, θ, and ,α can be interpolated from the set of measured transfer functions as shown in FIG. **8**. For example, it is known that the sound pressure from an audio source is inversely proportional to distance in normal atmospheric conditions. Furthermore, a transfer function of a point in space can be approximated by the weighted average of the two nearest measured transfer functions. FIG. **8**. shows an example of an approximate transfer function $H(i_3, jw)$ interpolated in the transverse plane at point i_3 from transfer functions $H(i_1, jw)$ and $H(i_2, jw)$, which are also interpolated from transfer functions $H(m_1, jw)$ and $H(m_2, jw)$ measured with speakers **#1 (89)** and **#2 (90)**.

Thus;

$$H(i_3, jw)=[H(m_1, jw)+H(m_2, jw)]/[2*L_{at}(jw)] \quad [1]$$

where L_{at} (jw) is the atmospheric loss transfer function due to atmospheric absorption and spreading roll-off of sound.

Similarly, interpolation can be used to approximate any transfer function at an arbitrary point in a three-dimensional space from the weighted average of the nearest set of measured transfer functions. The accuracy of interpolated functions can be improved if additional measurements are made with additional speakers and/or speaker-head orientations. The preferred embodiment of the invention employs a practical compromise between the number of speakers, e.g. six in the embodiment of the invention described herein, and individual orientations, e.g. two: a front and a back orientation. Furthermore, non-linear weighting for transfer function interpolation may be more appropriate if determined from statistical data obtained from transfer function measurements of large number of individuals.

Other transfer functions measured by the VEA system include:

(1) the $H_{icp\text{-}rec}$ (jw) transfer function, which represents the ICP receiver to in-the-ear-canal electroacoustic transfer function, as measured by a probe when the ICP is positioned in the ear canal of the individual;

(2) the $H_{icp\text{-}mic}$ (jw) transfer function, representing the electroacoustic transfer function from an ICP speaker to the microphone of the hearing aid used during the hearing aid evaluation; and

(3) the $H_{icp\text{-}fb}$(jw) transfer function, representing the acoustic leakage, i.e. acoustic feedback, from the receiver of the ICP measured at face-plate of the ICP.

The transfer functions $H_{ua}(p_n, jw)$, $H_{fp}(p_n, jw)$, $H_{icp\text{-}rec}$ (jw), $H_{icp\text{-}mic}$ (jw), and $H_{icp\text{-}fb}$(jw) are employed in various combinations to digitally synthesize acoustic signals, representing unaided, simulated aided, or aided listening conditions, with realism that is not possible with conventional evaluation and fitting methods.

In FIG. **9**, for example, a teacher-talker **101** and a child-listener **102** acoustic environment **100** is created as follows: direct acoustic paths p_{R1} and p_{L1}, and reflection paths p_{R2} and p_{L2}, for right and left ears of the child-listener **102** are represented by transfer functions interpolated from previously measured transfer functions of the child.

The acoustic realization of the environment of FIG. **9** is shown in FIG. **10**, in which a digital audio file **107** that represents teacher-talker speech is retrieved from a system memory **106** and digitally processed by digital signal processor **114**. The digital signal processor performs signal processes $H_{ua}(p_{R1}, jw)$ **108**, $H_{ua}(p_{L1}, jw)$ **110**, $H_{ua}(p_{R2}, jw)$ **109** and $H_{ua}(p_{L1}, jw)$ **111**, which represent the paths p_{R1}, p_{L1}, p_{R2}, and p_{L2}, respectively. Right and left ear path processes are summed at summing nodes **112** and **113** and are further processed with inverse transfer functions, $1/H_{icp\text{-}rec}$-Rt (jw) **(116)** and $1/H_{icp\text{-}rec}$-Lt (jw) **(104)**, for right and left ICP receivers **119/120**, respectively.

The inverse transfer functions are provided to cancel the acoustic transfer function that occurs between the ICP receiver and the residual volume of the ear canal as the sound is delivered. The processed right and left digital signals are then converted to analog signals via a digital-to-analog converter **115** and routed to right and left ICPs via an audiometric interface circuit **117**. The process of projecting a virtual audio image to a listener at a particular point in a three-dimensional space, such as teacher-talker speech to a child-listener, is referred to as spatialization.

Alternatively, live-voice signals from the operating clinician via the operating clinician microphone can be used, instead of digital audio data, for spatialization and delivery to the listener wearing the ICP pair. The virtual position and volume of the spatialized audio source are under the control of the virtual audiometer system of the present invention, as is explained in more detail below.

Transfer function measurements of linear time-invariant systems, such as the transfer functions H_{ua} (p_n, jw), H_{fp}(p_n, jw), $H_{icp\text{-}rec}$ (jw), $H_{icp\text{-}mic}$ (jw), and $H_{icp\text{-}fb}$(jw), typically employs discrete or swept pure tone acoustic stimulus. Other stimuli include speech-noise, white-noise, and other speech-like noise signals. Pseudo-random noise sequences and other signals have also been used to reduce the time required to compute the transfer function. Computational methods include Fast Fourier Transform (FFT), Maximum-Length Sequence (MSL), and Time-Delay Spectrometry (TDS) (see Rife. D., Vanderkooy, J., *Transfer-Function Measurement with Maximum-Length Sequences,* J. Audio Engineering Soc., Vol. 37, No. 6, June 1989, pp. 418–442). The advantages of MSL and TDS measurement include reduction of room reflection effects on the transfer function. One important component of measured transfer functions used in the present invention is the direct path transfer function.

In the preferred embodiment of the invention, the VEA's probe microphones are calibrated at the head reference point when the VEA is first installed in its clinical setup. These calibration data, stored in the system memory, are subsequently used during transfer function measurements to correct for the unique frequency response characteristics of each probe microphone used and the unique characteristics of room acoustics.

FIG. **11** is a partially sectioned, perspective view showing an intra-canal prosthesis (ICP) for an ICP-ITE representing hearing aids for shallow ear canal placement; FIG. **12** is a partially sectioned, perspective view showing an ICP for an ICP-ITC representing hearing aids for deep ear canal placement; FIG. **13** is a perspective view showing an ICP face-plate end, including face-plate probe tube holders and probe tube placement; FIG. **14** is a partially sectioned, side view showing an ICP core module for a two-part ICP configuration; FIG. **15** is a partially sectioned, side view showing adjustable vent inserts for an ICP-ITE; FIG. **16** is a partially sectioned, side view showing an ICP-ITC sleeve for a two-part ICP configuration; FIG. **17** is a partially sectioned, side view showing a complete two-part ICP-ITC assembly; FIG. **18** is a partially sectioned, side view showing an ICP having a programmable vent; FIG. **19** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including direct acoustic coupling via a magnetic attraction method; FIG. **20** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including direct acoustic coupling via an acoustic coupler method; FIG. **21** is a partially sectioned, side view showing a hearing aid and direct acoustic coupling method to an ICP, including a programming and acoustic coupling interface; and FIG. **22** is a partially sectioned, side view showing a hearing aid and acoustic coupling to an ICP via an acoustic coupler tip, all according to the invention.

In the foregoing figures, those elements of the invention that are common to the various embodiments have a common numeric designator. For example, the ICP of FIGS. **11** and **12** each have a receiver **136**, while the housing **129** in the embodiment of FIG. **11** is different from the housing **152** of the embodiment of FIG. **12**.

The intra-canal-prosthesis (ICP), shown in FIGS. **11–22**, consists mainly of a receiver **136**, a receiver port **199**, a probe tube **133** inserted in probe tube canal **134**, vent inserts **128** inserted in vent canal **130**, a probe microphone **131**, a face plate **122**, and a housing made of a flexible material, such as an acrylic. The ICP is generally designed to represent physical and electroacoustic characteristics of a desired type of hearing aid with the exception of the signal processing and generation, which is performed by the audio synthesizer board of the computerized virtual electroacoustic audiometer system. FIGS. **11** and **12** show ITE and ITC ICPs that represent hearing aids having shallow and deep canal placement, respectively.

The receiver **136** used in the preferred embodiment of the present invention (manufactured by the Knowles Corp. of Itasca, Ill.) was chosen for its acoustic characteristics, which are similar to receivers used in commercially available hearing aids, as well as its very low noise output characteristics. ICP receiver variations from simulated hearing aid receivers are stored in the VEA system memory as a correction transfer function used during various simulation processes. The probe tube **133**, preferably made of a silicone rubber material and having a diameter of approximately 1 mm, is inserted in the probe tube canal **134** of the ICP as shown in FIGS. **11–22**.

A vent canal **130** is preferably provided for pressure equalization in the ICP-ITC versions that have deep canal insertion depths (FIGS. **12** and **17**), and to accommodate vent inserts for the ICP-ITE version having shallow canal insertion depths (FIGS. **11** and **15**). In the ICP-ITE versions, a vent canal allows the insertion of various vent inserts into the vent canal to achieve desired in situ acoustic characteristics. For example, a vent insert of relatively large diameter may be used to reduce the occlusion effect that results from increased perceived volume of the individual's own voice. On the other hand, a smaller vent insert may be used to eliminate acoustic leakage from the receiver via the vent insert. A miniature connector socket **138** and connector plug **123** electrically connects the ICP to the VEA system via attached connector cable **125**.

The VEA system, in conjunction with the probe microphone system, permits measurements of the occlusion effects versus ICPs and vent types, as is explained later. The ICP also contains two probe tube holders **124** and a placement handle **126** for placement of the probe tube, as shown in FIGS. **11**, **12**, and **17**. FIG. **13** shows a more detailed illustration of a face plate **122**, including the face plate tube holders **124**. In the figure, a ICP/ITC sleeve **156**, and a hearing aid microphone position **132** are also shown. This configuration is used when measuring acoustic leakage feedback and face-plate transfer functions.

The ICP housing (**129**, FIG. **11**; **152**, FIG. **12**) is preferably made of a soft flexible material with acoustic baffling effects to provide comfort and acoustic sealing. Several versions of the ICP can accommodate a variety of ear canal sizes. For example, a small housing version is more suitable for pediatric populations, while a larger version is suitable for adults who have large ear canals. The ICP, shown in FIGS. **11** and **12** is preferably disposable to avoid contamination from individuals who have infected ear canals.

An alternate embodiment of the invention provides a two-part ICP configuration, as shown in FIGS. **14–17**. A core part **169** (FIG. **14**) is inserted in a variety of disposable sleeves **177**, as shown in FIGS. **15** and **16**. This option provides an economical alternative to the configuration shown in FIGS. **11-13** because only the sleeve component is disposable. The core part **169** is encapsulated in a protective material, preferably having semi-flexible properties. A decoupling capacitor **167** may be used to filter extraneous electromagnetic signals that cause audible noise.

The sleeve part shown in FIGS. **15** and **16** is typically made of flexible material, such as a soft acrylic, such that the ICP fits comfortably into a variety of ear shapes and sizes. FIG. **16** shows a sleeve suitable for deep canal insertions, representing ITC and CIC hearing aid types. Also shown in

FIG. **16** is an acoustic baffle system **186** that provides an acoustic seal while the ICP is inserted in the ear canal.

FIG. **15** shows an ICP sleeve for shallow canal insertions representing ITE hearing aid types. The ICP core is inserted in the sleeve cavity **179** of any ICP, including those shown in FIGS. **15** and **16**. The specific size of the ICP sleeve selected by the operating clinician depends upon the test performed, individual canal size, and hearing aid simulation requirements. An example of the combined parts of a core ICP and an ICP sleeve are shown in FIG. **17**, which represents an ICP-ITC assembly.

FIG. **18** shows a variation of the vent mechanism where the size of the vent is electronically controlled and adjusted (see Zdeblick, K., *A Revolutionary Actuator For Microstructures,* Sensors Magazine, eb. 1993). This is accomplished by employing programmable micro-valve **193** (such as the N0-300 manufactured by Redwood Microsystems of Redwood City, Calif.) which contains a silicon diaphragm **194** which is to regulate the size of the vent attached to the vent canal **197** via the micro-valve port **195**. Typical vent size range is between 0.032 and 1.5 mm, according to the voltage level supplied from the virtual electroacoustic audiometer module in response to operating clinician test selections.

The ICP is also used in a novel way to test a new type of hearing aids adapted to interface to the ICP, as shown in FIGS. **19–22**. Unlike conventional hearing aid and aided hearing evaluation methods that typically employ remotely positioned speakers to deliver acoustic signals into the hearing aid microphone, the ICP of the present invention presents acoustic signals directly to the microphone **211** of the hearing aid **214**. The acoustic coupling of the present invention spans a minimal distance typically less than 15 mm.

FIGS. **19** and **21** show an embodiment of the invention in which acoustic coupling is accomplished via a magnetic attraction method. In such method, the ICP receiver **136** is coupled to the hearing aid microphone **211** via magnetic attraction between a magnet disk **206** on the receiver end of the ICP and another magnet disk **209** near the hearing aid microphone port **210**, and which is part of the face-plate **218** of a hearing aid **214**, as shown in FIG. **19**. A sealing ring **205** provides acoustic sealing to minimize leakage in the coupling. Also provided are a hearing aid battery holder **221**, a hearing aid volume control **219**, a hearing aid circuit **212**, and a hearing aid vent canal **217**, all representing conventional components of a hearing aid device.

Additionally, the embodiment of the invention shown in FIG. **21** provides a programmable hearing aid circuit **253** that allows dynamic ITE testing via control signals routed from the VEA over a programming cable **257**. FIG. **21** shows an electrically programmable hearing aid with a programming cable **257** connecting the hearing aid circuit to the VEA of the present invention. These hearing aids contain circuits that are programmable or adjustable, typically via electrical signals. The shown programming interface at the face-plate is via the battery holder which is adapted to route programming electrical signals to the hearing aid circuit. The programming signals and interface methods are typically unique to the hearing aid model as provided by the specification of the hearing aid circuit used. These programming signals and interface methods are known to persons skilled in the art of hearing aid design. Other programmable hearing aids currently commercially available employ ultrasonic or infra-red signals with the appropriate signal interface circuits within the hearing aid.

An alternative acoustic coupling method couples the ICP receiver **136** to the hearing aid microphone **211** via a acoustic coupler **243**, as shown in FIG. **20**. The extended microphone port **242**, unique to the present invention, also acts as a handle to facilitate insertion and removal of hearing aid **214** during its normal use.

Another embodiment of the invention, shown in FIG. **22**, employs an acoustic coupler **290** adapted for insertion into a microphone port **299** of the hearing aid **214**. The microphone port **299** is recessed to accommodate an acoustic coupler tip **291**.

Another acoustic coupling method (not shown) employees a suction-cup ring to couple the ICP receiver to existing conventional hearing aids that are not equipped with special interface parts.

One major advantage of the direct acoustic coupling of the present invention is to improve the signal-to-noise ratio at the microphone of the hearing aid while the aid is being adjusted or evaluated. This is primarily accomplished by acoustically isolating the microphone of the hearing aid from ambient room noise via its coupling to the ICP.

Hearing aids of the present invention also employ a probe tube canal to allow for probe tube insertion and subsequent in-the-ear-canal acoustic measurements via the probe measurement system as shown in FIGS. **19–22**. The conventional method of in-the-ear-canal measurements with hearing aids involve probe placements beneath the hearing aid which subjects the probe to pinching effects, thus affecting the accuracy of the measurement. Furthermore, placing the probe tube beneath the hearing aid creates an acoustic leakage path which causes oscillatory feedback. The probe tube canal of the present invention also provides an improved method of advancing the probe while the hearing aid is placed in the ear canal.

The sequence of these phases as outlined in FIG. **23** represents a typical fitting process unique to the system of the present invention. The fitting process offered by the virtual electroacoustic audiometer system in the preferred embodiment of the present invention is implemented in five phases: (1) reference measurements **264**, (2) unaided hearing evaluation **265**, (3) predicted aided evaluation **266**, (4) simulated aided evaluation **267**, and (5) aided evaluation **268**. However, individual phases or a components of each phase can be administered individually, or in other sequence as suitable for the individual under hearing evaluation. Each process phase is implemented in a graphical module, as shown in FIGS. **24–28**.

The first phase, i.e. reference measurements, is implemented by a reference measurements module (FIG. **24**) that contains a reference measurement window (shown open in FIG. **24**) and a signal model window (shown iconized in FIG. **24**). The reference measurement window allows for measurements of various transfer functions that are used later throughout the fitting process.

The unaided transfer function H_{ua} (p_n, jw) described above, is measured when the 3D-REUR (3 Dimensional Real-Ear Unaided Response) option is selected.

Measurements are obtained from the frontal (facing speaker #1) or back (facing opposite speaker #1) orientations, depending on the Front/Back option selected. Plots of right and left ear transfer functions can be displayed in either transverse or sagittal plane depending on the Transverse/Sagittal option selection. FIG. **24** shows a set of 8-paired H_{ua} (p_n, jw) transfer functions in the transverse plane. The measurement is performed by positioning the individual centrally to the speaker array (discussed above) and placing right and left probe tubes in their respective unoccluded ear canal.

Another novel feature of the invention is the ability to measure and quantify the occlusion effect of the simulated hearing aid, as well as the fitted hearing aid.

However, before the occluded measurement is taken, a reference measurement with the ear canal unoccluded must be taken. The procedure, briefly described here, is to request the individual to utter a vowel, preferably a vowel with high energy contents in its low frequency spectrum, such as "ee." A measurement is taken with the probe positioned near the tympanic membrane. The occlusion effect reference measurement, i.e. unoccluded, is saved for occlusion effect measurement with the ear canal occluded using either the ICP or the hearing aid, as is explained below. The occlusion effect reference measurement is performed when the occlusion reference option is selected.

The face-plate transfer function H_{fp} (p_n, jw) (plots not shown) is measured when the Face-Plate Response option is selected. The ICP is placed in the ear and the probe tube tip is placed in the microphone position **132** of the face-plate as shown in FIG. **13**.

The ICP-receiver to real ear transfer function, $H_{icp\text{-}rec}$ (jw) is measured when ICP Calibrate option is selected. This requires the probe tube to be inserted in the probe tube canal of the ICP, and the tip of the tube near the tympanic membrane.

To facilitate the proper placement of the probe in the ear canal during various response and calibration measurements, a novel method is employed to optimize such probe placement within the ear canal, and specifically to minimize the effects of standing waves present in the ear canal due to wave reflections from the tympanic membrane. The frequency dependent standing wave patterns are well characterized and known to persons skilled in the art of acoustics and particularly real ear acoustic measurements. The new method of the invention involves acoustic presentation of a dual tone, one at a low frequency in the range of 1 kHz to 5 kHz, and a second at a range of 15 kHz to 20 kHz. The acoustic response to tone signals delivered either via a speaker or the ICP receiver, depending on measurement, is continuously measured by microphone probe system and displayed on the monitor, as shown in FIGS. **30–32**.

A plot of the acoustic response in an ear of an individual for each tone, shown in FIG. **29**, indicates a characteristic rise in the low frequency response and a notch in the high frequency response as the probe is advanced closer to the tympanic membrane. This notch occurs at approximately 5 mm from the tympanic membrane for the 15 kHz tone. Monitoring of the relative response characteristics during probe insertion provides a visual and computer-assisted method to indicate proper probe positioning as shown in the spectrum plots of FIGS. **30–32**. The end of this procedure is generally indicated when a significant notch, typically exceeding 15 dB as shown in FIG. **31**, followed by a significant rise in the high frequency, ie. second tone, response.

The low frequency, i.e. second tone, response shows only a small increase, within 3 dB, as the probe is inserted closer to the tympanic membrane. Although probe tip to tympanic membrane distance approximation is possible with this procedure, the object of this procedure is to position the probe such that minimal standing waves are present at frequencies of interest during transfer function measurements. For example, if unaided response measurements up to 6 kHz are desired, advancing the probe until detecting a notch in 15 kHz response ensures measurement errors not to exceed 2.5 dB at 6 kHz. Improved accuracy can be achieved by selecting a higher frequency for the second tone, although this increases the chance of advancing the probe too far, resulting in touching the surface of the tympanic membrane, an occurrence that is generally safe but that may cause discomfort.

Other combinations of tones, including a single, triple, composite, and other signals can also be used to implement the above procedure of continuously measuring the response to various acoustic stimuli and detecting an appropriate stopping point during probe advancement, with little regard to probe distance to the tympanic membrane. The appropriate probe position is referred to hereafter as the probe reference point.

The second phase, unaided evaluation, is implemented by an unaided evaluation module, shown in FIG. **25**, which consists of an unaided analysis window, shown open in the figure; a spatialization window, also shown open; a signal model window, shown iconized; and an audiometric evaluation window, also shown iconized.

The unaided analysis window allows for various in-the-ear-canal measurements and displays for hearing evaluation in the unaided condition while the ICP is inserted in the ear canal. Measurements and plots include Audiogram spectrum, Distortion, Time Analysis, Spectrogram, and 2-CC curves. Acoustic stimuli, measurement methods, and associated plots for these tests are known to persons skilled in the arts of audiology and signal analysis. However, the Audibility Spectrogram is a new feature that is unique to the present invention as described below.

The Audibility Spectrogram is a spectral plot showing the audibility of a signal with respect to the hearing profile of the individual and the critical audibility features of an acoustic signal. The audibility spectrogram is essentially a three-dimensional matrix represented in a two-dimensional plot that indicates signal dynamics (time) and Critical Audibility Regions (CAR) versus frequency, as shown in FIG. **25**. CARs, shown as the outer contours, are specific to each signal segment that is selected from the signal model window. CARs of a speech segment are defined by the critical sound features, such as the energy of significant formats in vowels, the energy of fundamental frequency of voicing, the energy of aperiodic frequency sounds, and other criteria known to effect intelligibility, detection, or identification, depending on the signal model selected.

The Audibility Spectrogram plots are derived by combining spectrograms of analyzed signals and defined CARs, and probe measured spectrograms computed and compared with the measured hearing profile of the individual at the CARs. Measured spectrogram values that fall below the threshold of hearing for the individual are assigned to Below Threshold (B-Thresh) values which define the outer contour region, within the CAR; while measured spectrogram values that exceed the threshold of hearing within CAR are assigned Above Threshold (A-Thresh) values which define a region within the Below Threshold region; and measured spectrograms values that exceed the uncomfortable loudness level (UCL) of the individual are assigned Above-UnComfortable Loudness level (A-UCL) values which define the inner-most contour regions.

The resulting color-coded plot is typically contour shaped for speech signals.

However, any type of acoustic signal can be assigned CARs and a corresponding audibility spectrogram based on the individual's measured hearing profile. The objective of the Audibility Spectrogram plot is to provide a quick graphical means of indicating the audibility of dynamically received acoustic signals by taking in consideration the individual's hearing profile and the critical audibility features of a signal model. This plot is particularly important in hearing aid fitting optimization processes during predicted aided, simulated aided, and aided evaluation.

The spatialization window permits selection of signal presentation mode, either in Spatialized or Intracranial

modes. Spatialized mode presents selected sources and background signals to be delivered to both ears via inserted ICPs according to the selected spatial relationship of head, sources, background, and boundaries, as shown in FIG. **25**. Spatial relationships include the distance between the audio source and the head reference point (d), azimuth angle (θ), and altitude angle (α).

Various individual and calibration transfer functions are employed to synthesize audio signals with realistic listening effects. Signal sources and corresponding levels are selected from the Signal Model window (not shown). Intracranial mode, on the other hand, offers the conventional sound presentation method where selected signals and corresponding levels are delivered without spatialization to one or both ears.

The Signal Model window permits the selection of source and background signals and corresponding level. Source selection may be of pure tone type, speech, music, or any signal of audiological significance. Background signals are typically competing speech, environmental noise, and other signals of audiological significance. The level of signals selected in the spatialized mode is preferably in dB SPL calibrated to 1 meter from the source in free field. The measured in-the-ear-canal acoustic response is preferably displayed in dB SPL as measured by the probe microphone system.

In the intracranial mode, source and background signals are routed to right, left, or booth ears as in conventional audiometry. The level of signals selected in the intracranial mode is preferably in dB SPL. The $H_{icp\text{-}rec}$ (jw) transfer function measurement via the ICP calibration procedure described above permits level selection in dB SPL. Furthermore, measurements via the probe microphone system can be made as needed to ensure that the probe and the ICP remain properly positioned in the ear canal.

A specific selection of source and background signal type, levels, and spatialization mode is defined as a signal model. One or more signal models can be selected, saved, and retrieved by the system for presentation and analysis purposes. A signal model can represent any individual or a combination of acoustic signals/scenarios, including speech, background noise, music, pure tone, masking noise, composite signals, and other audiologically significant signals.

The audiometric evaluation window, shown iconized, allows for various conventional audiometric measurements to be taken. This includes threshold audiogram, most comfortable level (MCL), uncomfortable loudness level (UCL), speech reception threshold (SRT), and various other audiometric measures known to persons skilled in the art of audiology. However, unlike conventional audiometry where transducers are calibrated in various acoustic couplers and measurements are measured in relative hearing level (HL) terms , the preferred method measures the in-the-ear-canal response in absolute sound pressure level (SPL) terms.

Another feature of the invention relates to the modes of audiometric signal presentation. As described above, spatialized or intracranial listening modes selected from the Spatialization window, not only affect the presentation selected from the Signal Model window, but also the Audiometric Evaluation window as well. For example, a standard audiological word list such as NU-6 or W-22, commonly used in conventional speech audiometry, can be presented in the conventional intracranial mode, or alternatively, in the spatialized mode unique to the invention.

The signal process of a spatialized unaided evaluation involves the unaided transfer function H_{ua} (p_n, jw), interpolated based on selections of the spatialization window, and the $H_{icp\text{-}rec}$ (jw) transfer function. A signal process implementation of a particular spacialized unaided evaluation is shown in FIG. **10**.

The third phase, the predicated aided evaluation, is implemented by the predicated aided evaluation module. This module, shown in FIG. **26**, allows the operating clinician to select a hearing aid and predict its performance without the involvement of the hearing-impaired individual. The module consists of a Hearing Aid Select/Adjust window, shown open; a Predicated Analysis window, shown open); a Signal Model window, shown iconized; a Spatialization window, shown iconized;and the Audiometric Evaluation module. The Signal Model, Spatialization, and Audiometric Evaluation windows are essentially identical to those described in the Unaided Evaluation phase.

The Hearing Aid Select/Adjust window permits hearing aid selection and subsequent adjustment. The predicated results of the selection/adjustment are shown on the selected plots of the adjacent Predicted Analysis window. Hearing aid selection can be automatic or manual, depending on the hearing aid selection Automatic/Manual option selected. Automatic selection involves selecting one or more hearing aids based on the fitting algorithm selected, and various other criteria selected by the hearing-impaired and the operating clinician. Conventional fitting formulae and methods, such as POGO, Berger, and NAL-II, are provided.

The preferred fitting method is the dynamic audibility method which employs a rational such that Audibility Spectrogram is optimized. This corresponds to plots that maximize the Above-Threshold (A-Thresh) contour areas while minimizing Bellow-Threshold (B-Thresh) and Above-UnComfortable loudness Level (AUCL) contour areas. Hearing aid models that best match the selected criteria are automatically retrieved from the system memory.

Alternatively, manual selection can be made by choosing one or more hearing aid models from the available list of models. A hearing aid model contains all of the necessary electroacoustic parameters that are used for signal processing of a signal model. The results of the signal process are used in the Predicted Analysis window for analysis and plotting purposes. Hearing aid parameters of a selected hearing aid model are adjusted automatically or manually depending on the hearing aid adjustment Automatic/Manual option and the fitting method selected.

A hearing aid control parameter set is typically unique to the hearing aid model selected. In the example window shown in FIG. **26** with hearing aid model DigiLink **100** selected, the control parameters are: volume control (VC), Low Frequency Cut (LFL), compression Threshold Knee (TK), Microphone type (MIC), Receiver type (REC), and Vent Size selection which reflects vent size of the ICP inserted. If a different vent size is selected, either manually via the vent insert selection, or electronically via the programmable micro-valve vent selection, a new $H_{icp\text{-}spkr}$ (jw) transfer function is preferably measured to improve the accuracy of the analysis.

The predicted analysis window is essentially identical to the unaided analysis window, described above, with the exception of the signal processing model that includes the measured face-plate transfer function H_{fp} (p_n, jw) (**292, 293**; FIG. **33**), hearing aid transfer function H_{ha} (jw) (**294**; FIG. **33**), and the measured ICP receiver to real-ear $H_{icp\text{-}rec}$ (jw) transfer function for the aided ear (**295**; FIG. **33**). The hearing aid H_{ha}(jw) transfer function is typically non-linear and varies depending on the hearing aid selected. The total hearing aid transfer function $H_{ha\text{-}t}$ (jw) typically includes transfer functions of the microphone H_{mic}(jw), hearing aid

circuit $H_{ha\text{-}rec}$ (jw), and the receiver $H_{ha\text{-}rec}$ (jw). The transfer function H_{ha} (jw) differs from $H_{ha\text{-}t}$ (jw) by excluding the hearing aid receiver and, instead, including a receiver correction transfer function $H_{Rec\text{-}corr}$(jw), that defines the difference between the predicted hearing aid receiver and the ICP receiver employed. This correction transfer function $H_{Rec\text{-}corr}$ (jw) is typically a linear transfer function and is supplied by the VEA system.

The predicted aided analysis process for an aided right ear and unaided left ear for a child-listener/teacher-talker scenario is shown in FIG. **33**. The results of the digital signal process are stored in the system memory **106** for analysis and display.

The analysis of the predicted data in the system memory includes audibility analysis as described above. The plotting includes an Audibility Spectrogram that indicates audibility contours of Below-Threshold, Above-Threshold and Above-UCL with respect to critical audibility regions (CRAs). FIG. **26** shows improved audibility in the predicted aided condition versus unaided condition shown in FIG. **25**, ie. increased Above-threshold contour areas.

Another prediction measurement unique to the present invention, is the measurement of occlusion effect caused by the insertion of the ICP into the ear canal that is characterized by the perceived amplification of the person's own voice. The present invention provides a method of measuring, subjectively and objectively, the magnitude of the occlusion effect. The subjective method is performed by asking the individual wearing the ICP to evaluate his own voice when speaking. If the response is objectionable to the hearing-impaired candidate then an alternative ICP, representing a different hearing aid, may be considered.

The objective method involves the measured response via the probe system in the occluded ear canal and subtracting the occlusion effect reference measurement, ie. uoccluded ear-canal measurement, as described above.

The patient microphone **57**, external to the ear canal, is typically employed to record the individual's own voice during occlusion effect measurements to ensure constant intensity level during both unoccluded and occluded ear canal measurements (see Mueller, H. G., Hawkins, D. B., Northern, J. L., *Probe Microphone Measurements: Hearing Aid Selection and Assessment,* 1992, pp. 221–224). A unique feature of the present invention is to eliminate not the only requirement of constant voice intensity, but also constant voice spectral characteristics. This is accomplished by adjusting the calculated occlusion effect measurement by the difference in the spectral characteristics of the individual's own voice.

It is known in the field of audiology that deep hearing aid insertion substantially reduces the occlusion effect, particularly at low frequencies in the range of 125 to 1000 Hz. Therefore, a smaller ICP, representing a smaller simulated hearing aid, may be used for subsequent evaluation phases.

The occlusion effect created by two types of ICP, i.e. ICP-ITC and ICP-ITE, is shown in the plot of FIG. **27**. This plot indicates a significant occlusion effect due to the ICP-ITE versus the ICP-ICP for an individual. This is expected since the ICP-ITE creates a greater residual volume, to which the occlusion effect is known to be directly proportional.

The advantage of ICP measurement at the probe reference point is that all measurements taken are independent of the ICP selected or its placement in the ear canal. However, to present accurate spatialized sounds to the individual, the $H_{icp\text{-}rec}$ (jw) transfer measurement is required whenever a new ICP is selected and inserted into the ear canal of the individual.

Another measurement unique to the invention is that of acoustic feedback caused by acoustic leakage from the ICOP receiver, when simulating a hearing aid receiver, to the face-plate of the ICP, which simulates the face-plate of the hearing aid. The transfer function $H_{icp\text{-}fb}$(jw) (**338**; FIG. **37**), e.g. amplitude and phase response, is measured at the face-plate as described above. The opening created by the removal of the probe tube from the ICP probe tube canal is preferably plugged during the feedback measurement to exclude acoustic leakage due to the probe canal.

A significant application of the feedback transfer function is in the simulation, and thus prediction, of oscillatory feedback of the simulated hearing aid. This undesirable oscillatory feedback manifests itself in the form of whistling, which interferes with the normal operation of the hearing aid. The prediction and simulation of the oscillatory feedback of a simulated hearing aid having a selected setting is accomplished by incorporating the ICP feedback transfer function $H_{icp\text{-}fb}$(jw) **337**, as shown in FIG. **37**.

Oscillatory feedback can be audible to the individual wearing the ICP via the ICP receiver. The oscillatory feedback can also be measured via the ICP microphone system in conjunction with the VEA system. This feature allows the operating clinician to adjust the settings of the simulated hearing aid, particularly the gain, frequency response, and vent size, such that oscillatory feedback is minimized or eliminated. Similarly, the VEA system can be employed to select automatically an alternate hearing aid or alternate hearing aid parameter set, such that oscillatory feedback is minimized or eliminated.

The predicted aided analysis window also includes other analysis and corresponding plots of Audiogram, Distortion, Time Analysis, Spectrogram, 2-cc Curve. These are standardized measurements and plots that are known to persons skilled in the art of hearing sciences and technology. The 2-cc coupler curves involve conversion of measured in-the-ear-canal response to standard 2-cc coupler curves using real-ear-to-2-cc coupler conversion formulas. Standard signal models, such as pure tones, are typically involved in the 2-cc coupler measurements (see *Specification of Hearing Aid Characteristics,* ANSIS3.22-1987, American Standards National Institute). Other evaluation methods conceived and well within the means of the invention include the Articulation Index (AI) measures for unaided, predicted aided, simulated aided, and aided conditions.

An objective of the predicted aided module is to predict objectively the performance of a selected hearing aid according to the selected signal model, selected hearing aid parameter set, and the individual's hearing profile, without the involvement of the hearing-impaired individual.

The fourth phase, simulated aided evaluation, is implemented by the simulated aided evaluation module, as shown in FIG. **27**. This module allows the operator to select and optimize one or more hearing aids and simulate their audible characteristics. The module consists of a Hearing Aid Simulation window, shown open; a Simulated Aided Analysis window, shown open; a Signal Model window, shown iconized; a Spatialization window, shown iconized; and the Audiometric Evaluation module, shown iconized. The Signal Model, Spatialization, and Audiometric Evaluation windows are essentially identical to those described above. The Simulation Aided window is essentially identical to the Hearing Aid Select/Adjust window of the Predicted Aided Evaluation module. Similarly, the Simulated Aided analysis window is essentially identical to the Predicted Analysis window.

A major difference in the simulated aided evaluation module is the module's ability to synthesize simulated aided

conditions and to present the audible results to the hearing-impaired individual. Another significant difference is that analysis is performed by the module based on measured, rather than predicted, data. The measured response is obtained via the microphone probe measurement system with the probe tip placed at the probe reference point, as discussed above.

An example of a simulated aided signal process, shown in FIG. **34**, involves the transfer function of the hearing H_{ha} (jw) that includes the $H_{Rec\text{-}corr}$ (jw), and the face-plate transfer function H_{fp} (p_n, jw) for simulation of the aided ear. The results of the process are converted to analog signals via the digital-to-analog-converter **115** and routed to the right and left ICPs, **119** and **120** respectively, inserted in the ear canals of the individual.

If the microphone of the predicted hearing aid is of directional type, then separate microphone transfer functions, representing its directional properties are employed, as shown in FIG. **35**. A digital audio file **107** is retrieved from the system memory **106** and processed with face-plate transfer functions $H_{fp}(p_1,jw)$ (**310**; FIG. **35**) and H_{fp} (p_2, jw) (**312**; FIG. **35**), where p_1 and p_2 represent two points in a three-dimensional space. Signal paths from p_1 and p_2 may represent direct and primary reflective paths, respectively. Secondary reflective paths p_3, p_4, . . . , p_n (not shown) can be similarly represented in the digital signal process.

The results of each face-plate transfer function step are further processed with the corresponding microphone transfer function **318**, **320** for each signal path from points p_1, p_2, . . . p_n. The results are summed **326** and are processed by the hearing aid circuit transfer function $H_{ha\text{-}cir}$ (jw) **322**, $H_{Rec\text{-}corr}$ (jw) **324**, as shown in FIG. **35**. The resulting digitally processed signal is then converted to analog signal via the digital-to-analog converter **115** and routed to the appropriate ICP within the ear canal via the audiometric transducer interface **117**.

The simulated aided analysis window includes measurements and corresponding plots of Audiogram, Distortion, Time Analysis, Spectrogram, Audibility Spectrogram, 2-cc Curve, Occlusion Effects, and Feedback Analysis. These measurements are essentially identical to those described above for the predicted analysis window. This process is based on the system's ability to compute a hearing aid prescription based on a selected fitting prescription formula/rational. The selected hearing aid can be adjusted and results analyzed and plotted with or without the involvement of the hearing-impaired individual.

An objective of the simulated aided module is to optimize, objectively and subjectively, the performance of a selected hearing aid according to measured in-the-ear-canal probe response as a function of the selected signal model, hearing aid parameter set, the individual's measured hearing profile, and subjective responses to the presented audible signal.

One feature unique to the invention is the ability to compute the characteristics of a simulated monaural or binaural hearing aid system that produces natural sound perception and improved sound localization ability to the hearing impaired individual. This is accomplished by selecting a simulated hearing aid transfer function that produces, in conjunction with the face-plate transfer function, a combined transfer function that matches that of the unaided transfer function for each ear. The matching requirement typically involves frequency and phase responses. However, the magnitude response is expected to vary because most hearing impaired individuals require amplification to compensate for their hearing losses.

Once the hearing aid selection and optimization processes are completed via VEA system simulation, the characteristics of the simulated hearing aid are translated to hearing aid specifications for manufacture/assembly.

Manufacturing specifications include: hearing aid components simulated by the VEA system, including the microphone and receiver; shape and size of hearing aid according to the ICP selected; hearing aid circuit blocks and circuit components; hearing aid parameter setting; and vent type/size. An objective of the VEA system is to provide a detailed specification to the manufacturer/assembler to manufacture/assemble a monaural or binaurally matched hearing aid system that closely matches the preferred simulated hearing aid. Ordering of the actual hearing aid is performed from the Order menu shown in FIG. **27** which provides a printout of detailed hearing aid specification.

The final step in the process, aided evaluation, is represented by the aided evaluation module as shown in FIG. **28**. This module consists of an Aided Evaluation window, shown open, an Aided Analysis window, shown open; an Audiometric Evaluation window, shown iconized; a Signal Model window, shown iconized; and a Spatialization window, shown iconized. The latter three windows are essentially identical to those in the predicted aided evaluation and simulated aided evaluation windows. The aided evaluation window permits electronic adjustment of manufactured hearing aid parameters as in the case of a programmable hearing aid, shown in FIG. **21**, or displaying the suggested parameter setting in the case of a manually adjusted hearing aids, shown in FIG. **20**.

The aided analysis window is similar to the analysis window for unaided, predicted aided, and simulated aided evaluation process steps, except that the measurements and corresponding plots reflect the response from the actual hearing aid inserted in the ear canal of the individual rather than predicted or synthesized signal, i.e. simulated aided, response analysis.

Synthesized realistic acoustic signals are presented to the hearing aid by coupling spatialized sounds directly to the microphone of the hearing aid, as shown in FIGS. **19–21**. The face-plate transfer function, H_{fp} (p_n, jw), and the supplied ICP receiver-to-microphone transfer function $H_{icp\text{-}mic}$ (jw) are employed in the digital synthesis process, as shown in FIG. **36**. A digital audio file **107** representing an audio source at location p_n in space is retrieved from the system memory **106** for processing with the free-field to face-plate transfer function H_{fp} (p_n, jw) **340**, **342** for right and left ears, individually. Other parallel processes reflecting filtering of additional audio sources or filtering of reflective paths, shown collectively in the dashed rectangles **341**, **343**, are summed with the right **112** and left **113** summing nodes. The outcome of summing nodes is further processed to equalize the ICP receiver to hearing aid microphone coupling effects by applying the inverse transfer function $1/H_{icp\text{-}mic}$ (jw) **344**, **345**. The acoustic signals supplied to the microphones **350** of the hearing aids **351** represent spatialized signals with characteristics selected and controlled by the VEA system operator via the Spatialization, Signal Module, and Audiometric Evaluation windows.

Electroacoustic testing of the hearing aid, coupled with the ICP as described above, may also be performed external to the ear canal, for example 2-cc coupler measurements can be performed by connecting the receiver output of the hearing aid to the 2-cc coupler input. The ICP, in conjunction with the signal generation capability of the VEA, can produce various acoustic stimuli as input to the hearing aid during its 2-cc coupler-based hearing aid evaluation.

Similarly, 2-cc coupler measurements can be performed on the ICP, i.e. a simulated hearing aid, by connecting the receiver output of the ICP to the 2-cc coupler input.

The invention not only deals effectively with today's diagnostic and fitting problems but also provides a basis for new tools that are audiologically significant. For example, the system's ability to synthesize realistic acoustic conditions, both simulated aided and aided, can be used as an auditory rehabilitative tool where a hearing impaired listening ability is improved by interactive training. In such application, the hearing impaired person is presented with spatialized signals that represent spoken words in noisy background. Even though the words might be audible as determined from the audibility measurements and methods described above, these words might not be intelligible for the untrained hearing-impaired individual. Depending on the verbal response, or registered response via a response keypad, the VEA system can provide audible or visual feedback to the hearing impaired individual that indicates the appropriateness of the response. The object of this new test is to teach the hearing-impaired how to improve speech perception and intelligibility beyond mere audibility.

Another test made possible by the invention determines the individual's ability to localize a sound in a plane or in three-dimensional space. An example is the detection of minimal audible angle (MM) test whereby the individual's ability to detect, in degrees, the minimal angular separation of pure tones versus frequency (see Mills, A. W., *On the Minimum Audible Angle,* Journal of Acous. Soc. of Am. 30:237–246, 1956). Furthermore, a comparison of the individual's localization ability can be compared across unaided, simulated aided, and aided conditions.

The invention also makes it possible to determine the individual's ability to detect sound movements in a plane or in a three-dimensional space. For example, a sound object can be synthesized to represent movement in a particular geometrical and frequency pattern. The individual's impaired ability to detect the movement can be assessed. Furthermore, a comparison of the individual's ability to detect sound movements can be compared across various listening conditions in the unaided, simulated aided, and aided conditions.

Although the invention is described herein with reference to the preferred embodiment, one skilled in the art will readily appreciate that other applications may be substituted for those set forth herein without departing from the spirit and scope of the present invention. Accordingly, the invention should only be limited by the Claims included below.

I claim:

1. A method for selecting and presenting binaural acoustic stimuli in spatialized mode for hearing diagnostics and rehabilitation, comprising the steps of:

synthesizing audio signals, including primary audio signals comprising pure tones and speech, and secondary audio signals comprising background noise and other competing sources, wherein said synthesized audio signals are presented according to individualized transfer functions that include a hearing aid faceplate, and any of an individual's effects of body, head, and ear on incoming signals;

controlling spatialization parameters of said audio signals, including any of the position of each source in space in terms of any of distance, azimuth, and altitude; and acoustic boundary parameters including room size, reflection properties, reverberation, atmospheric absorption, and spreading loss roll-off;

presenting such spatialized stimuli to an individual; and

performing any of hearing diagnostics, hearing aid prescription, hearing aid simulation, and hearing aid fitting with an audiometric module.

2. The method of claim **1**, further comprising the step of:

presenting such spatialized stimuli to predict performance of one or more hearing aid systems.

3. The method of claim **1**, further comprising the step of:

presenting such spatialized stimuli for simulating a hearing aid system.

4. The method of claim **1**, further comprising the step of:

presenting such spatialized stimuli for hearing aid evaluation.

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